PROGRAM OF THE

Seventy-Fourth Meeting of the Acoustical Society of America

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TUESDAY, 14 NOVEMBER 1967 — SILVER CHIMES EAST, 9:00 A.M.

Session A. Ultrasonic Visualization I: Methods, Instrumentation, and Illustrative Examples.

(Joint technical session ASA/American Institute of Ultrasonics in Medicine)

WILLIAM R. TURNER, *Chairman*

Invited Papers (25 minutes)

A1. Ultrasonic Two-Dimensional Visualization for Medical Diagnosis. GEORGE KOSSOFF (nonmember), *Commonwealth Acoustic Laboratories, Sydney, Australia.*—In ultrasonic two-dimensional visualization, the spot size of the display unit is the limit of resolution and determines the design of the echoscope. The shortest echo response is obtained from a nonreflectivity backed transducer, but the usual backed or quarter-wave matched transducer gives adequate axial resolution. The beamwidth of the optimum weakly focused transducer is equivalent to four times the spot size, and some form of signal comparison must be used to make the azimuthal resolution compatible with the axial resolution. The finite beamwidth also degrades the axial resolution of inclined structures and distorts the display by elongating structures along the beamwidth. Off-axis receiving transducers may be used to receive larger echoes from inclined structures. An improvement by a factor of 3 may be obtained before the employment of these transducers degrades the resolution. The technique is prone to artifacts that are more troublesome in contact than in water delay scanning. These features are illustrated for an echoscope designed for the visualization of the pregnant uterus.

A2. Omnidirectional Scanning and Relief Presentation for Ultrasonic Visualization of Intracranial Anatomy. WILLIAM J. FRY, *Interscience Research Institute, Champaign, Illinois, 61820.*—The application of new scanning (omnidirectional), echogram composition (band assembly), and display (relief presentation) methods to the ultrasonic visualization of intracranial anatomy is described. These methodologic advances are implemented by a general purpose on-line digital computer that makes possible a flexible approach to the development of a more sophisticated generation of such instrumentation facilities. At this stage of the evolution of instruments for viewing the brain, it is desirable to separate certain problems specifically associated with transskull operation from those concerned with determining what anatomic features can be detected and localized when examining pulses are employed in the absence of enhancement of impedance differences at interfaces and under the condition that the acoustic energy not traverse bone. Results for rhesus monkey brain *in vivo* under such experimental conditions are illustrated.

A3. Evaluation of Intensity-Modulated Recording for Ultrasonic Diagnosis. JOHN M. REID, *Department of Physiology and Biophysics, University of Washington, Seattle, Washington, 98105.*—An evaluation of current intensity-modulated recording systems has shown that they will not produce identical records from the same echo pattern. Differences in the records are due to two factors: the recording transfer function, and the ways in which the zero signal level or "baseline" is established and maintained. To find a correlation between ultrasonic echo patterns and the underlying tissue structure, it is essential to be aware of the changes in records that are due to the recording equipment. Lack of knowledge of a particular equipment can have a disastrous effect upon attempts to duplicate the results of another investigator. Differences in the transfer functions of current equipment arise from the use of different types of cathode-ray tubes and recording film as well as the use of a video differentiator. An additional effect of not maintaining a constant zero level is to increase the variability of the amplitude of received echoes. Several tests have been devised by which the processing inherent in a particular type of equipment can be assessed and by means of which the changes introduced by different combinations of characteristics can be demonstrated.

A4. Symmetrical Scanning of the Head with Ultrasound. D. M. MAKOW (nonmember), *Division of Applied Physics, National Research Council, Ottawa, Canada,* AND D. L. MCRAE (nonmember), *Department of Radiology, University of Toronto, Toronto, Canada.*—The absolute position determination of reflecting targets in the brain is difficult because of the refraction of the ultrasonic beam and the different velocities in the bones of the skull and brain. Since many brain disorders distort the almost symmetrical brain structures, emphasis should be placed more on the detection of asymmetries rather than on faithful mapping of the brain. The asymmetry can be displayed with considerable precision by symmetrical scanning of the head simultaneously from both sides with two transducers. An ultrasonic immersion scanner that employs this technique will be described, and the results will be discussed.

A5. Performance Limitations on the Ultrasound Image Converter. JOHN E. JACOBS (nonmember), *Biomedical Engineering Technology Institute, Northwestern University, Evanston, Illinois 60201.*—In recent years, the ultrasound image converters have become available in such a form as to lend themselves to routine day-to-day use. The systems are capable of resolving detail in the order of a few hundred microns at sound levels incident of 10^{-7} W/cm^2. The greatest difficulty presently associated with the systems is the need for liquid immersion of the object being examined. It is the purpose of this discussion to delineate the various factors controlling the performance of the system and, through motion-picture film, show the application of the image system to typical problems in medicine and biology.

A6. Holographic Imaging with Ultrasound. FREDRICK L. THURSTONE, *Biomedical Engineering, Duke University, Durham, North Carolina.*—Ultrasound fields, having both spatial and temporal coherence, have been used to produce sonic holograms that can produce three-dimensional images in a visible light field. The sound field is detected by mechanically or electronically scanning a piezoelectric element and determining both magnitude and phase of the sonic field. The information thus obtained is processed sequentially by electronic, optical, and photographic techniques to produce a hologram transparency. The generation of the images by wavefront reconstruction is accomplished with the coherent light produced by a laser.

TUESDAY, 14 NOVEMBER 1967 — EMPIRE ROOM, 9:00 A.M.

Session B. Psychological and Physiological Acoustics I: Calibration, Thresholds, and Noise

BERTRAM SCHARF, *Chairman*

Contributed Papers (10 minutes)

B1. Use of an Adapter Ring for Calibration of Large Circumaural Earphones. LAURA ANN WILBER, JOHN SWINDEMAN (nonmember), ANN HOGUE (nonmember), AND VICTOR GOODHILL, *Department of Surgery, University of California, Los Angeles, California.*—Routine calibration of large circumaural earphones has long been a problem due to lack of a standard coupler, variability found with placement on a brass plate coupler, and difficulties in probe tube calibration techniques. In order to provide a practical solution to the above problems so that a pair of large circumaural (Pedersen) earphones could be used routinely, four studies were carried out at UCLA. In these studies, a Plexi-glass adapter ring was developed that, when fitted with a standard artificial ear and attached to the earphones, furnished a method whereby routine acoustic calibration of the earphones could be carried out easily and repeatedly. Comparisons were made between the Pedersen earphones and TDH 39 earphones using: (1) the adapter ring and artificial ear; (2) a probe tube; and (3) psychophysical techniques. These comparisons made it possible to achieve a method for routine calibration of the large circumaural earphones. The above studies also indicated some problem areas in the use of certain psychophysical techniques such as loudness balancing.

B2. A Calibration Study of One Hundred Audiometers Currently Employed in Hearing Conservation Programs. W. G. THOMAS AND MACK J. PRESLAR, *Department of Surgery, University of North Carolina, Chapel Hill, N. C. 27514.*—This study describes results based on the calibration of 100 audiometers over a 3-yr period. The audiometers tested were in current general use from 11 different agencies or individuals, and represented 30 different models from eight manufacturers. The audiometers were calibrated with high-quality laboratory-standard equipment under the most careful conditions. Sound-pressure output levels of the earphones were recorded for each frequency and attenuator setting from maximum output to 20-dB hearing loss (HL). Some difficulty was encountered with acoustic measurements below 30-dB HL using ISO standards. Below this level, terminal voltage readings were used. Only two audiometers could be judged in calibration when strict compliance with the standards was observed. The most frequent problem found was incorrect sound-pressure output of the earphone. Eighty-nine of the 100 audiometers failed to meet this specification. The second most common problem involved excessive rise time. The third most common problem was frequency outside the tolerance limit. This was followed in order by failure to meet specifications for hearing-loss interval and hearing-loss range, instability of SPL output with ac line voltage variations, excessive harmonic distortion, and excessive shock hazard. [Work supported in part by U. S. PHS National Center for Chronic Disease Control.]

B3. Acoustic Wavemotion in the Human External Ear. E. A. G. SHAW AND R. TERANISHI, *Division of Applied Physics, National Research Council, Ottawa, Canada.*—The pressure distributions generated in an external ear replica by a nearby point source have been measured with a probe microphone. Five normal modes have been clearly identified, M1 at 2.9 kHz that is essentially the $\lambda/4$ resonance of the canal proper with a large end correction, M2 at 5.5 kHz the fundamental depth resonance of the concha with large radiation damping and weak canal coupling, M3 at 9.3 kHz related to the first transverse resonance of the concha, M4 at 11.2 kHz, and M5 at 12.8 kHz a pair of modes related to the second transverse resonance. M1 and M2 are consistent with limited measurements on real ears and are easily reproduced in a simple geometrical model. Above 6 kHz, transverse wave-motion in the concha is largely responsible for large variations of response with angle of incidence. Real ears, replica, and model all exhibit relatively low on-axis response in the 8-kHz region.

B4. Influence of Polarity of Acoustic Stimulus on Psychophysical Threshold. J. L. HALL, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—A 3-msec burst of a 1-kHz tone was added to every eighth cycle of a continuous 50-Hz tone and presented monaurally through a TDH-49 earphone. The threshold intensity of the high-frequency burst was determined as a function of two parameters [Deatherage *et al.*, J. Acoust. Soc. Am. **38** (1965)]: (a) intensity of the low-frequency tone (110, 100, 90, and 80 dB *re* 0.0002 dyn/cm^2, and low-frequency tone absent); and (b) phase of the low-frequency tone at which the high-frequency burst occurred (increments of 45°). At the three highest intensities, changes in phase produced appreciable threshold shifts, repeatable from subject to subject. The results are interpreted in terms of the relationship between polarity of displacement of the basilar membrane and initiation of activity in the auditory nerve (Kiang *et al.*)

B5. BUDTIF Thresholds: Variability and Bias. RICHARD A. CAMPBELL, *Case Western Reserve University, Cleveland, Ohio, 44106.*—Threshold determination with the block up-and-down, two-interval forced-choice (BUDTIF) method has been investigated by obtaining 10 repeated 75% correct thresholds from human subjects. Major parameters were number of trials per block (2, 4, and 8) and number of trials per threshold (16–144). Measures of within- and between-day variability and within-run variability were closely related to one another and were generally inversely related to block size and number of trials per run. The effect of number of trials on variability with a block size of four trials was minimal however. For example, the mean between-day standard deviation was 0.80 for 32 trials and 0.73 for 144 trials. Mean thresholds were directly related to block size, as one would expect from the binomial bias. Thresholds also tended to become lower with increased trials per run. Efficiency was generally inversely related to trials per run and directly related to block size, although the trends are somewhat mixed. The variability observed in these data differ from that observed in stimulated listeners. This is interpreted to indicate that the assumption of trial-by-trial independence in human observers is much less tenable than commonly asserted.

B6. The Effect of Depth and Content of the Middle Ear Cavity on Underwater Hearing Thresholds. JOHN F. BRANDT, *Communication Sciences Laboratory, University of Florida, Gainesville, Florida, 32601.*—The audibility threshold performance of the human ear submerged in water at ear depths of 12, 35, 70, and 105 ft is compared to its performance in air. Threshold SPL's at 125, 250, 1000, 2000, and 8000 Hz from six divers wearing open circuit, air breathing SCUBA equipment were obtained by the Békésy technique. Differences between water and air conduction thresholds ranged from about 30 dB SPL *re* 0.0002 μ at 250 Hz to 60 dB at 2000 Hz. Water conduction thresholds that are generally independent of test frequency require about 71 dB SPL. No significant effect upon threshold was provided by ear depth. Thresholds were also obtained from three divers at 105 ft breathing compressed air and, in another condition, breathing a mixture of helium and oxygen. Significantly higher thresholds (about 5 dB) at all frequencies were obtained with the helium–oxygen mixture in the middle-ear cavity. [Research supported in part by Office of Naval Research and National Institutes of Health.]

B7. Sound-Shadow Effect in Impulse Noise. DAVID C. HODGE AND R. BRUCE MCCOMMONS (nonmember), *U. S. Army Human Engineering Laboratories, Aberdeen Proving Ground, Maryland 21005.*—The sound-shadow effect of the human head in an impulse-noise field was studied by exposing 27 subjects to gunfire noise so their left ears were normal to the oncoming shock wave (near ear) and their right ears were protected by the shadow of the head (far ear). Noise exposure was continued until the subject's near ear demonstrated 15 dB TTS and the postexposure TTS in near and far ears was compared. Peak pressure level at the entrance of the far-ear canal was less than one-half that found at the near ear (153 vs 161 dB *re* 0.0002 μbar). Mean TTS was significantly smaller in the far ears than in the near ears. The mean "protection" afforded the far ear by the head's shadow ranged from 3 dB at 1 kHz to 12 dB at 6 kHz. The implications of the findings for the protection of weapon crewmen and others is discussed.

B8. Discriminations Among Simulated Sonic Booms. J. W. LITTLE (nonmember), *The Boeing Company, Seattle, Washington.*—A waveform generator was constructed that provided for the introduction of sounds represented by four different waveforms into a box 21 in. square, 25 in. high, and with an 18-in. speaker mounted in the top of the box. The four waveforms were an N wave, a differentiated N wave, one-half square wave, and a differentiated one-half square wave. With their heads sealed in the small box, 16 subjects made discriminations among the six different pairs of stimuli. That the subjects could not discriminate between an N wave and a differentiated N wave, or between a square wave and a differentiated square wave, reproduces the results of Zepler and Harel (1965), but with different equipment and psychophysical technique. Using earphones, their experiments showed that the loudness of the N waves is determined by the two rising parts (pulses), while our experiment shows that when the pulses are identical in amplitude change and direction, discrimination is not possible. That subjects could discriminate among the four pairs of stimuli where compression was compared to rarefaction is probably caused by the nonlinear characteristics of the ear, but could also result from air leakage at the neck seal.

B9. Loudness and Annoyance of Clicks. HERMAN R. SILBIGER, *Bell Telephone Laboratories, Inc., Holmdel, New Jersey 07733.*—This investigation was concerned with the subjective effects of high level impulses of short duration, to add to previously reported data [H. R. Silbiger, J. Acoust. Soc. Am. 38, 937(A) (1965)]. The principal parameters under investigation were the click shape, duration, and peak pressure level. The last ranged from 98 to 159 dBt. Subjects gave magnitude estimates of the loudness of the clicks and rated the subjective disturbing effects on an annoyance scale. The developed annoyance scale appears to be linear with positive peak pressure level. The slope of the average loudness function was 0.8. Data on the difference limen for these clicks will also be reported. Because of the level of the stimulus clicks, the subjects' thresholds were continuously monitored, and temporary threshold shift curves were obtained for both single and multiple click presentations. These data again show evidence of both positive and negative TTS, with the amount of negative TTS decreasing as the number and level of stimulus clicks increases.

B10. Loudness and Annoyance of Impulsive Office Noise. ERNEST H. ROTHAUSER, *IBM Research Laboratory, Zurich, Switzerland,* AND GUNTHER E. URBANEK AND W. PACHL (nonmember), *Institute for Telecommunications, Technical University, Vienna, Austria.*—A study has been made on the evaluation of natural impulsive noises by subjective and objective procedures. One special example was used to get an indication of whether existing objective calculation methods might give sufficient agreement with subjective tests. The office noise sample studied bursts from a single typewriter operated at repetition rates between 1.5 and 15/sec. A forced-pair comparison procedure of the constant stimulus type was used for the subjective evaluation. This choice has been prompted by the good results that we obtained with a corresponding method for speech quality measurements. In the subjective tests reported here, we asked for loudness and for annoyance comparisons. Two types of broad-band noise were utilized as reference signals. In addition to this, narrow-band noise centered at 1000 Hz was used to calibrate the two broad-band noises subjectively, and was also compared directly to the test noise. The subjective measurement results of this study are compared with those obtained by several well-known objective methods for noise rating and loudness evaluations.

B11. Threshold Shift in Hearing as a Function of Bandwidth and Mode of Noise Exposure. ALEXANDER COHEN AND

EMANUEL JACKSON, JR. (nonmember), *National Center for Urban and Industrial Health, Public Health Service, Cincinnati, Ohio*—Fifteen subjects were each given separate 15-min exposures to three bandwidths of fatiguing sound (whole octave band and $\frac{1}{3}$-oct band centered at 1000 Hz, pure tone of 1000 Hz) presented in three modes (constant level, variable level, and intermittent). The $\frac{1}{3}$-oct band SPL were 5 dB below the whole octave-band levels for the various exposure conditions that, according to the CHABA criteria, should have yielded equivalent threshold shifts (TTS). The pure-tone SPL's equaled the $\frac{1}{3}$-oct band levels for one set of exposures, and were 3 dB lower in another set. These latter conditions were intended to verify the increased noxiousness of pure-tone versus narrow-band noise stimulation, also specified in the CHABA criteria. Comparisons of the TTS produced by the different bandwidths and modes of noise presentation suggested that the CHABA criteria may error on the conservative side, especially in rating the hazards to hearing of $\frac{1}{3}$-oct band and variable types of noise exposure. Implications of these findings as regards critical-band-TTS notions and the action of the acoustic reflex are noted.

TUESDAY, 14 NOVEMBER 1967 — MEDALLION WEST, 9:00 A.M.

Session C. Noise I

T. F. W. EMBLETON, *Chairman*

Contributed Papers (10 minutes)

C1. Noise Reduction Studies for a Multistage Axial Flow Transonic Compressor. DAVID CHESTNUTT (nonmember), *NASA Langley Research Center, Hampton, Virginia.*—It is the purpose of this paper to describe an aircraft-type transonic compressor built especially for noise research studies and to present some preliminary results. This compressor is unique in that it is adjustable over a very wide range of operating conditions. Initial test results indicate that the relative-tip Mach number is a dominant factor in noise generation. Results of systematic measurement studies for a range of pressure ratios, rotational speeds, and inlet flow angles suggest that variation of in-flow angle may be an effective technique for markedly reducing inlet noise radiation during high-tip speed operation.

C2. Theoretical Analysis of Compressor Noise. M. V. LOWSON, *Research Staff, Wyle Laboratories, Huntsville, Alabama 35807.*—The noise radiated by rotor–stator and stator–rotor interactions in a duct of negligible length has been analyzed with the aid of the author's general theory. Explicit analytic results for rotating fluctuating source patterns have been found, which reduce to the case of Gutin for the steady propeller case with no axial velocity, and to Garrick and Watkins' results for the propeller problem with axial velocity. Computed results of over-all power and directionality will be presented for several basic cases, and the practical implication of these theoretical results will be discussed. [This work has been supported by the National Aeronautics and Space Administration under contract.]

C3. Achieving Noise Reduction Economically. R. E. JELINEK AND K. S. NORDBY, *Systems Development Division, International Business Machines Corporation, Endicott, New York 13760.*—The noise control activity of an acoustics group at a development laboratory dealing mainly in computer peripheral equipment is presented. The cost of product noise control increases rapidly as machines progress in hardware development, becomes expensive when tooling is completed, and usually prohibitive after release to production. The goal, therefore, is to implement noise control early, i.e., on the designer's drawing board before hardware exists. However, to development engineering, noise control is only one of many factors to be considered within limitations of budget and schedule. Therefore, the acoustics group must have an effective program to ensure proper noise control emphasis during various phases of development. The program of the group is presented: before-hardware participation in machine development; recommendations of machine acoustical objectives based on experience and market requirements; prediction of noise-control measures required to make machines meet their acoustical objectives from design layouts and power-level measurements of components and subassemblies; evaluations of complete machines in a semianechoic chamber; interactions with industrial design, human factors, heat transfer, installation planning, product test, and manufacturing quality control groups.

C4. Analytical and Experimental Studies of Sound Pressures in Ducted Propellers. WERNER FRICKE AND JOHN R. BISSELL (nonmember), *Bell Aerosystems Company.*—Various types of VTOL aircraft and air-cushion vehicles under development utilize ducted propellers. Consequently, knowledge of the sound-field inside ducts is required to develop design criteria for lightweight duct structures of sufficient acoustic fatigue life. The paper presents an analytical approach for predicting the rotational sound due to blade loadings in propeller ducts, based on definable propulsion quantities. The sound field in the propeller plane is represented by expanding the rotating net propeller blade pressure distribution in a suitable series, being a solution of the wave equation that satisfies the boundary conditions associated with the duct and hub. The equations obtained allowed a calculation of the spectral and spatial sound-pressure distribution along the axis and radius. The validity of the analytical approach was investigated by exploring the sound field of a full-scale model duct under measured operational conditions. These experiments were conducted in the large-scale wind tunnel at NASA Ames Research Center. An additional experimental study was conducted on the effect of increased tip clearance on the sound pressure. [Program sponsored by NASA Headquarters.]

C5. Rotor–Stator Interaction in an Axial Flow Blower. T. F. W. EMBLETON, *Division of Applied Physics, National Research Council, Ottawa, Ontario, Canada.*—Measurements have been made on a ducted, single-stage blower, 12 in. in diameter, rotating at 4500 rpm. The duct includes a venturi, enabling mechanical power and efficiency to be measured simultaneously with SPL's. Recently, Lowson ["Reduction of Compressor-Noise Radiation," J. Acoust. Soc. Am. 40, 1248(A) (1966)] surveyed various possible noise-reduction methods. The experiments support qualitatively some of his estimates, though some quantitative differences also occur. As a function of axial spacing between rotor and stator, the SPL (both the pure-tone spectrum at the blade-passage frequency, and broad-band noise with the pure tones removed) shows a break in slope at a spacing rougly equal to the chord of a rotor blade. At smaller spacing the slope lies between 3 and 10 dB per doubling of spacing—depending on stator blade pitch, number of stator blades, and other parameters—while at larger spacing the sound levels decrease between 0 and 2 dB per doubling of spacing. Mechanical efficiency is independent of spacing up to about one chord length; at greater spacings it usually decreases slightly—typically it is reduced 2%–3% at a spacing of three or four chord lengths.

C6. Predicting Hearing Impairment from *A*-Weighted Sound Levels. James H. Botsford, *Bethlehem Steel Corporation, Bethlehem, Pennsylvania.*—The hearing deficiency of a group may be described in terms of the percentage having impaired hearing in order to focus attention on the abnormalities of medical significance. The prevalence of hearing impairment for speech as defined by the medical profession was determined for various populations using published hearing survey data. The incidence of hearing impairment was found to increase with age in all populations and also with *A*-weighted sound levels in the groups exposed to harmful noise at work. The effect of harmful noise exposure at work is to increase the incidence of hearing impairment above that found in unexposed populations of the same age. Curves were drawn for predicting the net increase in incidence of hearing impairment resulting from exposure to various *A*-weighted sound levels at work. The increased impairment predicted for noise exposures that do not exceed the limits currently recommended was found to be acceptably small.

C7. Predicting Speech Interference and Annoyance from A-Weighted Sound Levels. James H. Botsford, *Bethlehem Steel Corporation, Bethlehem, Pennsylvania.*—A sample of 741 noises measured in factories and in construction equipment was used to relate speech interference levels to *A*-weighted sound levels. A mathematical formula was developed which predicts the speech interference level from the *A*-weighted sound level with little error. A graph was drawn for use in predicting speech interference effects directly from *A*-weighted sound levels. Another sample of 235 noises measured along industrial plant boundaries was used to relate *A*-weighted sound levels to the level rank used for rating annoyance. A method was developed for predicting the level rank from the *A*-weighted sound level with satisfactory accuracy. A graph was drawn for predicting the degree of annoyance directly from the *A*-weighted sound level. The methods developed for predicting speech interference and annoyance effects directly from *A*-weighted sound levels are believed to be as accurate as the methods utilizing octave-band sound-pressure levels on which they are based.

C8. Recent Research Relative to Perceived Noise Level. R. J. Wells, *General Electric Company, Schenectady, New York.*—This paper is a progress report on a joint project involving the General Electric Company and Bolt Beranek and Newman Inc. Certain refinements in the computation of perceived noise level are under consideration. A new method of spectral integration of annoyance relative to the rating contours is examined together with possible revisions in the correction procedure employed for pure tones. Possible slight changes in the basic NOY rating contours will also be discussed. Correlations of jury tests with annoyance calculations are improved by these changes.

C9. Measurement of the Power Output of Pure Tone Sources in a Reverberation Chamber. George C. Maling, Jr., *Acoustics Laboratory, International Business Machines Corporation, Poughkeepsie, New York 12602.*—A true rms voltmeter (80-sec integration time) and a moving microphone have been used to measure the space-averaged mean-squared sound pressure ($\bar{p}^2$) in a reverberation chamber produced by pure tone sources and air handling devices that produce discrete tones. The data are in good agreement with previous results [IBM J. **11**, No. 5 (1967)]. The statistical stability of an estimate of $\bar{p}^2$ may be determined from $n=2l/L$ where n is the degrees of freedom of the estimate, l is the microphone path length, and L is the correlation distance for the sound pressure. For a pure-tone source, the change in power output W with source location in the chamber has been measured in the frequency range 500–2000 Hz, and has been found to be in good agreement with theory. For an air handling device, measurements show that the power radiated at the blade passage frequency and its harmonics are less dependent on source location than the output of an ideal (monopole) source. The statistical stability of an estimate of W has been determined as a function of frequency.

C10. Digital Implementation of a Noise Rating Meter and Transient Analyzer. William W. Lang, George C. Maling, Jr., and William A. Taylor (nonmember), *Acoustics Laboratory, IBM Corporation, Poughkeepsie, New York 12602.*—The "back-door" approach to noise level evaluations described by R. H. Peterson [J. Acoust. Soc. Am. **37**, 1197(A) (1964)] has been implemented on a general-purpose digital computer. The method is useful for evaluating the noise radiated by a machine or other source that produces single or repeated bursts. With this method, the octave-band spectrum and noise rating of a single machine cycle or other transient are determined after one or more segments of the signal are attenuated. This enables the noise control engineer to identify those portions of the transient that contribute significantly to the noise rating. The Cooley–Tukey algorithm is used to compute the power spectrum. The transient is segmented into short (8.192 msec) records so that the methods developed by Welch [IEEE Trans. Audio Electron. **15** (1967)] may be applied. Applications to transients produced by representative IBM machines are presented.

C11. Category Scaling Judgment Tests on Motor Vehicle and Aircraft Noise. Karl S. Pearsons, *Bolt Beranek and Newman Inc., Los Angeles, California.*—Tests have been conducted in which subjects were asked to rate the sounds produced by aircraft flyovers and motor vehicle drivebys on a category scale. Recorded sounds were rated by college students in an anechoic chamber; "real-life" sounds were rated by college students and community residents at locations near a highway and two airports. The subjective ratings were correlated with the calculated perceived noise levels of the sound. In the laboratory and field environments, the results indicate little difference in the ratings as a function of perceived noise levels. In addition, the relationship between noisiness rating and perceived noise level is quite similar among the various noise types tested. [Work supported by the Aircraft Development Service, Federal Aviation Administration.]

C12. Experimental Study of Aerodynamic Sound from a Rotating Disk. Robert C. Chanaud, *Purdue University, Lafayette, Indiana.*—The broad-band sound generated aerodynamically by a smooth disk rotating in its own plane was measured. The low-frequency part of the spectrum was caused by vortex formation at the disk edge, and appeared acoustically as a point pressure dipole on the disk axes and aligned with the axis. The high-frequency part of the spectrum was caused by the interaction, with the disk edge, of the boundary layers on both faces and on the periphery. The location of the effective source moved closer to the edge with increasing frequency, and was pressure dipole in nature. Addition of a sandpaper ring to both faces caused generation of high-frequency sound at the point of roughness as predicted by [J. Acoust. Soc. Am. **37**, 516–522 (1965)]. Acoustically, the source appeared as dipoles whose axes were about 70° to the axis. Edge and roughness caused sounds are exceptions to the concept that surface dipoles account only for reflection of quadrupole sources [Powell, J. Acoust. Soc. Am. **32**, 982–990 (1960)].

C13. Cross-Spectral Density of the Noise on a Baffle Beneath a Plane Covering Layer. S. GARDNER, *Hydrosystems, Inc., Farmingdale, New York.*—Analytical results have been obtained for the cross-spectral density of the noise pressure on an infinite plane baffle located beneath a fluid layer, when random pressure fluctuations present on the upper surface of the layer have spatial scales that are small in comparison with an acoustic wavelength, and the baffle acts as a "locally reacting" surface. (1) If the layer thickness is small in comparison with an acoustic wavelength, and large in comparison with the length scales of the surface pressure, the pressure spectral density on the baffle is inversely proportional to the layer thickness. The random pressure on the baffle is isotropic, with correlation lengths proportional to the layer thickness. (2) If the layer thickness is large in comparison with an acoustic wavelength, the pressure spectral density on the baffle is independent of the layer thickness. The random pressure on the baffle is isotropic, with correlation lengths proportional to the acoustic wavelength. [Work supported by the Office of Naval Research, Code 468.]

TUESDAY, 14 NOVEMBER 1967 — MEDALLION EAST, 9:00 A.M.

Session D. Physical Acoustics I

G. MAIDANIK, *Chairman*

Contributed Papers (10 minutes)

D1. Right-Angle Reflector Technique for the Ultrasonic Examination of Material Properties. FRED R. ROLLINS, JR. AND S. L. LEVY (nonmember), *Midwest Research Institute.*—An ultrasonic goniometer is described that utilizes a single transducer and a reflector composed of two plane orthogonal surfaces. With the instrument immersed in a suitable coupling liquid, the transducer rotates about an axis colinear with the intersection of the orthogonal surfaces. The amplitude of the reflected signal is automatically recorded versus the angle of incidence measured relative to one of the reflector walls (the removable specimen). Amplitude variations can be related to the excitation of surface waves, Lamb waves, critical refraction conditions for body waves, and other physical properties of the reflecting surfaces. The instrument has been extremely useful in measuring surface wave velocities for both single crystal and polycrystalline specimens. On copper and quartz single crystals, good agreement has been achieved with theoretical predictions of surface wave velocity as a function of propagation direction. [Phys. Rev. **119**, 533 (1960); J. Acoust. Soc. Am. **41**, 921 (1967)]. The sensitivity of the device to other physical parameters will be discussed. [Work supported in part by the Office of Naval Research].

D2. Propagation of Unstable Finite-Amplitude Waves. A. L. VAN BUREN, *Department of Physics, The University of Tennessee, Knoxville, Tennessee.*—When a finite-amplitude ultrasonic wave in water is reflected from an interface, a phase shift of the Fourier harmonic components can occur. The new phase relationship between the harmonics may cause the wave to be unstable. Depending on the phase shift, the amplitude of the harmonics may increase or decrease as the wave tends toward a stable configuration. A model that assumes that the reflection of these components is the same as if they were independent waves is used to calculate phases (and amplitudes) of these components as the wave propagates after reflection. These results are compared with experiment. (Stabilization distances are not sensitive to variatons in $\alpha_0 L$, where α_0 is the low-amplitude attenuation coefficient and L is the discontinuity distance. [Research sponsored by the Office of Naval Research.]

D3. Spectrum Produced by an Ultrasonic Resonant Cavity. M. A. BREAZEALE AND CHING-TU CHANG (nonmember), *Department of Physics, The University of Tennessee, Knoxville, Tennessee.*—A spectrum analyzer is used to examine the ultrasonic spectra produced when large-amplitude ultrasonic waves are caused to resonate in a water-filled cavity. The cavity, consisting of a plane transducer and concave reflector, produces frequency components that in some cases are neither integral submultiples, nor integral overtones, of the driver frequency. [Research sponsored by the Office of Naval Research.]

D4. Subharmonics Generated by Bubbles in Sound Fields. E. A. NEPPIRAS, *Department of Physics, University of Vermont, Burlington, Vermont.*—Measurements have been made of the acoustic output from isolated air bubbles in water and glycerol–water mixtures subjected to sound fields. A mechanical device has been developed for generating small bubbles of uniform size at a controlled rate. The bubbles are allowed to rise along the axis of a radial-focusing transducer capable of operating over a wide frequency range (10–60 kc/sec) at intensities up to the cavitation threshold. It is found that driven at twice their natural radial resonant frequency may emit a strong signal at half the excitation frequency. A weak emission at the subharmonic is observed even at low drive intensities. At a certain threshold, often well below the threshold for transient cavitation (estimated from white noise), the response suddenly increases, and may remain at a high constant level for many cycles. At the cavitation threshold, the output becomes intermittent, and other bubbles are formed and emit, along with a large increase in white noise. The indications are that nonlinear radial vibrations contribute most to the subharmonic response. [Work supported by a contract from National Institutes of Health, U. S. Department of Health, Education, and Welfare.]

D5. Bubble Growth Rates and Thresholds of Rectified Diffusion. ANTHONY ELLER, *Acoustics Research Laboratory, Harvard University, Cruft Laboratory, Cambridge, Massachusetts 02138.*—A gas bubble that ordinarily would dissolve may be seen to grow by rectified diffusion if it is driven by an acoustic field whose pressure amplitude exceeds a threshold value. This threshold pressure has been measured as a function of bubble radius for air bubbles in water that is saturated with dissolved gas. The acoustic frequency was 26.6 kHz. The rate of bubble growth or dissolution, when the acoustic pressure was not equal to the threshold value, was also measured. The experimental results are compared with predictions of theory.

D6. Gas Diffusion by Pulsating Bubbles in Sonic Fields. ROBERT K. GOULD, *Lafayette College, Easton, Pennsylvania.*—A pulsating gas bubble is suspended by an acoustic radiation pressure force in a liquid. The bubble is far from all boundaries and virtually stationary. When the sound field is relatively weak and the liquid is undersaturated with the gas in question, the bubble shrinks under the influence of normal diffusion of gas into the liquid. When the liquid is saturated and the sound field relatively strong, the bubble may grow or shrink under the opposing influences of surface tension pressure and sonic pressure (rectified diffusion). Measurement of the diameter of the bubble as a function of time permits, in the first case, determination of the diffusion constant for the gas-liquid system, and, in the second case, experimental evaluation of current theories of sonically induced rectified diffusion. It is found that when a bubble breaks into surface oscillations, whichever process it is undergonig at the time, either growth or decay, is enhanced. [This work was supported in part by NIH grant while the author was on sabbatical leave during the academic year 1966–67 at the University of Vermont, Burlington, Vermont.]

D7. Incipient and Desinent Threshold of Visible Cavitation in Helium II. R. T. BEAUBOUEF (nonmember), M. L. CHU (nonmember), A. MOSSE, AND R. D. FINCH, *Department of Mechanical Engineering, University of Houston, Houston, Texas 77004.*—Measurements have been made of the incipient and desinent thresholds of visible cavitation in helium II, between 1.8°K and the λ point. It was found that the incipient voltage threshold for visible cavitation was at least twice as great as the desinent threshold. The desinent threshold at the lower temperatures was an order of magnitude higher than the audible threshold, but become comparable to the audible threshold close to the λ point. A qualitative explanation of the existence of incipient and desinent thresholds is advanced on the basis of Hsieh's theory of rectified internal convection [Contract Nonr-Rept. No. 85-33]. Further experimental observations of the sizes and motions of visible cavities and of the noise spectra of cavitation are discussed in relation to this theory. [This work was supported by NSF grant.]

D8. The Damping of Perturbations on a Plane Shock Wave. M. G. BRISCOE (nonmember)* AND A. A. KOVITZ (nonmember), *Northwestern University, Evanston, Illinois.*—A plane shock front, propagating normal to itself through a uniform ideal gas, is given a sinusoidal perturbation at an initial instant of time. The subsequent motion of the front and the gas in its wake is analyzed using a linearized system of equations and auxiliary conditions. The analysis follows closely that due to Zaidel [J. Appl. Math. Mech. 24, 316–327] for a shock wave produced by a corrugated piston. Two cases are considered: (a) the normal shock reflection from a sinusoidal wall such that the usual wall boundary conditions are satisfied; (b) the free-space damping of a sinusoidal perturbation such that there be no disturbance on the backward propagating weak discontinuity emanating from the initial position and time of the perturbed shock. For sufficiently small shock Mach numbers the presence of the wall does not have a strong effect on the damping rate; damping is caused primarily by the direct interaction of the disturbed wake with the perturbed shock front. Experimental and theoretical results corresponding to Case a compare very favorably with each other. Case b has not been subjected to experimental study. The analysis may be extended to arbitrary initial shock shapes by Fourier integral or Fourier series techniques.

*Now at the von Karmen Institute for Fluid Dynamics, Brussels, Belgium.

D9. Profile of Repeated Shock Waves in a Tube. JAMES L. MCKITTRICK (nonmember) AND DAVID T. BLACKSTOCK, *Acoustical Physics Laboratory, E. E. Department, University of Rochester, Rochester, New York 14627*; AND WAYNE M. WRIGHT, *Kalamazoo College, Kalamazoo, Michigan 49001.*—Measurements of the waveform of a plane progressive sawtooth wave in a 2-in.-i.d. air-filled tube have been made with a wide-band condenser microphone mounted flush in the rigid end plate of the tube. The source, a horn driver at the other end of the tube, was excited with tone bursts to avoid standing waves. The frequency was 3.4 kHz and the sound-pressure level 154 dB. Under these conditions, the predicted shock formation distance is about 5 ft and the sawtooth distance about 8 ft. Measurements at 12 ft did not, however, confirm the expected perfect sawtooth shape. Although the bottom of each shock was sharp, the top was well rounded. Total delay between the bottom of the shock and the peak was about 25 μsec, i.e., about 10% of the wave period). At greater distances the delay was longer. The delay cannot be ascribed to poor microphone response. Tests with N waves in free space indicated a microphone rise time of less than 1 μsec. The rounding of the shock peaks apparently is caused by tube wall effects. [Work supported by NDEA, AFOSR, and ONR.]

D10. Characteristics of the Shock Wave Generated in Air by a Blasting Cap. LIPPE D. SADWIN (nonmember) AND ERMINE A. CHRISTIAN, *U. S. Naval Ordnance Laboratory, White Oak, Silver Spring, Maryland, 20910.*—The propagation of the shock wave generated in air by a very small surface explosion, a commercial No. 6 electric blasting cap, has been studied. A large number of time-resolved pressure histories were recorded with piezoelectric transducers at distances over the range of 10–265 ft from the explosion point. The observed range of peak air shock pressures was from 0.3 lb/in.^{2} down to 3×10^{-3} lb/in.^{2} Comparisons with TNT surface burst data for much larger explosions show that the peak pressure-distance attenuation exponents are the same. Based on these pressure-distance data, the yield of the No. 6 electric blasting cap used is approximately 0.5 g of TNT. Even at the lowest peak pressures measured, the wave shapes are characteristic of typical airblast shock waves. An essentially triangular positive pressure phase is followed by a negative pressure phase of much lower amplitude and longer duration. This is unlike the signatures of sonic booms that are highly symmetric. The effect of the asymmetry on the energy density spectrum of the blasting cap is also discussed.

D11. Near Noise Field of the Transitional Ballistics Phase. O. C. BIXLER, JR., H. E. DAHLKE, R. E. KAPLAN (nonmember), AND J. J. Van Houten, *LTV Research Center, Western Division, Ling-Temco-Vought, Inc., Anaheim, California.*—Transitional ballistics involves the events that occur at the muzzle of the weapon prior to, during, and immediately after the uncorking of the projectile. Shock waves precede and follow the projectile during this period of the ballistics cycle. Theoretical and experimental examination of this phase provides details regarding the near noise field associated with a weapon. Shadowgraphs of the nearfield shock waves and space-time diagrams for both supersonic and subsonic projectiles reveal the development of the acoustic and ballistic flow fields. The nonlinear nature of the near noise field is examined in detail by application of Whitham's theory of propagation in nonuniform channels. [Work supported by the Rock Island Arsenal, Illinois.]

D12. Transient Reflection of Acoustic Waves from Fluid Velocity Discontinuities. ALLAN B. FRIEDLAND, *AVCO Space Systems Division, Lowell, Massachusetts,* ALLAN D. PIERCE, *Mechanical Engineering Department, Massachusetts Institute of Technology, Cambridge, Massachusetts,* AND CHARLES C. MOO (nonmember), *AVCO Space Systems Division, Lowell, Massachusetts.*—A theoretical treatment is presented of the reflection of line source generated acoustic waves from an abrupt discontinuity in the ambient fluid velocity. The model employed is that of a homogeneous, compressible, isothermal fluid with negligible viscosity, thermal conductivity, and gravity effects, occupying infinite space. In the upper half region, the fluid is assumed to have a constant horizontal ambient fluid velocity, while in the lower half region, it is assumed at rest with a horizontal line source embedded in it. The time dependence of the line source is arbitrary. The method of Cagniard is employed to find a full-wave exact solution for the acoustic radiation in the lower-half space. Various anomalous features of the transient solution are discussed concerning their dependence on the fluid Mach number in the upper half space. The peculiar features of reflections at Mach numbers greater than 2, which was predicted by Miles and Ribner for incident plane waves, is discussed in some detail.

TUESDAY, 14 NOVEMBER 1967 — BURGUNDY EAST, 9:00 A.M.

Session E. Underwater Acoustics I: Propagation

A. O. WILLIAMS, *Chairman*

Invited Papers (25 minutes)

E1. **Total Reflection, Phase Changes, and Caustics.** IVAN TOLSTOY, *Hudson Laboratories of Columbia University, Dobbs Ferry, New York 10522.*—The simple theory and pertinent acoustic evidence, relating to the classic frequency-independent phase changes upon total reflection, are briefly reviewed. Of particular interest are experimental results showing the effect of such phase changes in modifying the shape of propagating pulses. Thus, it is well known that, in the plane wave approximation, if a narrow pressure pulse is totally reflected off the plane boundary of a high velocity medium, the reflected pulse may develop a tail. This is interpreted physically by the fact that a frequency independent phase change ϕ corresponds to a delay $\tau=\phi/\omega$, i.e., a delay inversely proportional to the angular frequency ω. The mechanism of tail formation is thus a delay. In the process of total reflection, it takes time for the energy to penetrate into the high-velocity medium and to be restored back into the medium of incidence. Of particular interest in both ocean and atmospheric acoustics is the case of total internal reflection in a medium of continuously increasing velocity. Standard asymptotic approximations, identical to those used in quantum theory, show that when certain conditions are fulfilled, an effective phase change of $\pi/2$ takes place. This of course implies the deformation of pulses upon total reflection, appearance of tails, etc. Some remarkably clear experimental evidence showing this effect has been obtained in recent years. It has sometimes been proposed that similar effects are predicted by the so-called phase changes at a caustic. This is not so. The subject of phase shifts at caustics and foci is well known to students of optics and diffraction theory. The theory shows that factors of $e^{i\pi/2}$ (caustics) or $e^{i\pi}$ (foci) make their appearance in the solutions describing the local field in the simple harmonic case. However, the mechanism by which a caustic or focus is produced is important in defining precisely the significance and limitations of the $e^{i\pi/2}$ or $e^{i\pi}$ factors in any given case, especially when applied to the spectra of broad-band disturbances traveling through the system. Thus, in the latter case, if one accepts the $\pi/2$ caustic phase shift at face value and applies it to a narrow pulse, one runs afoul of the causality principle. On the other hand, it is obvious that a caustic is simply a geometrical construct: The only thing that happens there is an interference between arrivals along neighboring rays, a phenomenon which cannot, in a linear approximation, leave any permanent imprint upon a pulse traveling through such a region. Thus, these "phase shifts" are localized effects, which cannot be applied indiscriminately to the spectra of pulses traveling along the edges of caustics or through foci. [Hudson Lab. of Columbia Univ. Informal Documentation No. *144*. This work was supported by the U. S. Office of Naval Research.]

Contributed Papers (10 minutes)

E2. **Focusing of Underwater Step-Waves near a Caustic.** ALEXANDER SILBIGER, *Cambridge Acoustical Associates, Inc., Cambridge, Massachusetts 02138.*—The pressure field near a caustic (convergence zone) is analyzed for transient pulses with vanishing rise time. On the caustic, linear acoustic theory predicts a singularity of the order of $(t-t_s)^{-1/6}$ at the shockfront, t_s being the arrival time of the front at the caustic. The corresponding total impulse remains finite. Factors limiting the peak pressure are discussed. Theoretical results are compared with data obtained by the Naval Ordnance Laboratory with explosive sources in a flooded quarry. [This work was supported by the U. S. Office of Naval Research.]

E3. **On the Existence of Focusing Zones along the SOFAR Axis.** PETER HIRSCH, *Bell Telephone Laboratories, Inc., Whippany, New Jersey.*—The typical SOFAR pulse is composed of many separate arrivals that have propagated along different, multiply refracted paths. The energy that propagates straight along the channel axis is called the axial arrival. Goodman and Duykers [J. Acoust. Soc. Am. 34, 960–962 (1962)] show that, if the sound-speed profile about the axis is approximately parabolic, then the axial arrival should be periodically focused at ranges that are integer multiples of a characteristic distance that is about 6 NM in the North Atlantic. However, it is known that the behavior of the paraxial arrivals is very sensitive to the precise form of the profile [see e.g. Hirsch and Carter, J. Acoust. Soc. Am. **37**, 90–94 (1965); Williams and Horne, J. Acoust. Soc. Am. **41**, 189–198 (1967)], so that the question of the existence of focusing zones in the ocean sound channel has remained open.

To answer the question, one can use a result first pointed out by Slichter [Physics 3, 273–295 (1932)], that focusing in space implies focusing in time. Thus, the nonexistence of axial focusing zones in typical SOFAR propagation can be inferred from the time dispersion of the arrivals. [Work supported by the Bureau of Ships, U. S. Navy.]

E4. Representation of SOFAR Speed by Spherical Harmonics. ROCKNE H. JOHNSON, *T Phase Project, Institute of Geophysics, University of Hawaii, Honolulu, Hawaii 96822.*—Accurate position fixing by SOFAR requires that sound travel times along the paths involved be accurately known. A general method for providing travel times along any path in a given region is by representing the SOFAR speed as a continuous function of latitude and longitude. Such a function has been constructed for the Pacific Ocean. The basic data were hydrographic measurements, from which local SOFAR speed data were obtained, together with explosion travel times, from which average velocities along particular paths were calculated. No allowance was made for seasonal variation, which might be predictable in high latitudes. The explosion data, which might be viewed as calibrations, have been obtained only in the northeast Pacific. Both types of data were combined in a single matrix equation to yield estimates of the coefficients of a set of spherical harmonic functions. Classical harmonic analysis was precluded by the mixed data as well as by its irregular spacing. Fougere's method of harmonic analysis, which orthogonalizes the condition matrix by the Gram–Schmidt process, was applied. Termination of the series at degree six left a standard deviation of the hydrographic data of 2.0 msec.

E5. Geometrical Acoustic Propagation with Frequency Dependence. L. P. SOLOMON AND D. K. Y. AI (nonmember), *Aerosciences Laboratory, TRW Systems, Redondo Beach, California.*—One characteristic of geometrical acoustic solutions is their frequency independence. Geometrical acoustics stems from the solution of the eikonal equation that is a form of the wave equation in the limit of infinite frequency. Since the effects of the medium and the boundary conditions on the acoustic fields are definitely frequency dependent, it is desirable to obtain approximate solutions that reflect this dependence on frequency. Although data are available on propagation losses for signals of different frequencies, geometrical acoustics does not offer a means of comparison. An asymptotic expansion in terms of inverse powers of the frequency is presented. The first term in the expansion obeys the eikonal equation; all the ray-path solutions obtained from geometrical acoustics are applicable. Here, the first term of the expansion, which is the geometrical acoustic field, is obtained in closed form using a simplified profile in which the first derivative is continuous everywhere. The calculation of the higher-order terms then involve operators that have known variable coefficients which are functions of the lower-order solutions.

E6. Ray Propagation through a Moving Nonhomogeneous Medium, with Linear and Exponential Variations. I. D. TRIPATHI, *Electrical Engineering Department, University of Houston, Houston, Texas 77004.*—Two simple models of the ocean are considered. In the first model, the body of water is assumed to be moving along the horizontal or x axis with a velocity that decreases linearly with depth. The speed of sound in this medium is assumed to also vary linearly with depth when the medium is at rest. In the second model, the water velocity along the x axis is assumed to decrease exponentially with depth and the speed of sound is assumed to vary exponentially with depth when the medium is stationary. The vector eikonal equation is used to find the ray path in these two ocean models, and the expressions for the ray path and its slope are derived and plotted. Previous work in this field is reviewed, and its scope is discussed. [This work was sponsored by the Acoustics Program Office of Naval Research under contract.]

E7. Study of the Effect of Plane Boundary Location in the Analysis of Characteristic Dispersion. F. R. DINAPOLI, D. J. DEFANTI (nonmember), AND F. H. MIDDLETON, *University of Rhode Island, Kingston, Rhode Island.*—The mathematical analysis of the characteristic dispersive effects of a continuously stratified ocean requires obtaining a solution to the reduced wave equation that simultaneously satisfies the existing boundary conditions. The relative ease with which the dispersion information can be extracted from this boundary value problem is influenced by the location of the boundaries. The desired eigen values for a finite ocean are often embedded in an intractable transcendental equation, the solutions of which are usually obtained through the application of numerical techniques. However, if the physical problem is approximated by an ocean of infinite extent, the characteristic values are given by an exact, explicit, and easily obtained relation. The purpose of this work was to study the effect of this approximate mathematical model upon the characteristic dispersion of the medium. This was accomplished by comparing results from both models for a number of both single- and double-valued sound velocity profiles. In both cases, the exact solution was sought in terms of hypergeometric functions. For the infinite ocean model, further restrictions were applied so that the solution might be expressed in terms of Jacobi polynomials. [Research supported by U. S. Naval Underwater Sound Laboratory, New London, Connecticut.]

E8. Normal Mode Solution in a Medium with Curvilinear Velocity Profile. S. M. HAMZEH (nonmember), *TRACOR, Incorporated, Austin, Texas 78701.*—The solution of the wave equation is obtained for a fluid medium with a curvilinear velocity profile. The reciprocal of the velocity squared is represented by an nth-degree polynomial as a function of depth. This leads to depth-dependent special functions numerically evaluated from recurrence relations by digital subroutines. The general solution is expressed as an integral in the wavenumber space. The integrand is analytic in the cut complex wavenumber plane except at a countably infinite number of poles that characterize the normal modes of propagation. The residue theorem transforms the integral expression to an infinite series, the normal modes contribution, plus a branch line integral that vanishes in the long-range limit. In the short-range limit, the branch line integral leads to a wave diminishing in amplitude as the inverse square of the range and having a phase velocity equal to the speed of sound in the ocean's bottom.

E9. Calculation of Underwater Sound Intensities by Numerical Integration Techniques. MAX K. MILLER, *Texas Instruments Inc., Dallas, Texas.*—Calculation of sound intensities in the ocean where the velocity profile is given at discrete values of water depth presents difficulties. Suitable functions must be used in order to fit the measured points, and then be integrated to give correct sound intensities. Overlooked is the fact that when the measured velocity is given at discrete points, calculations lend themselves naturally to numerical integration techniques. In this paper, Chebyshev quadratures and Simpson's one-third rule are used to calculate sound intensities by integrating expressions containing the velocity function. These methods avoid the problem of fitting the velocity profile with a function that can be integrated to yield meaningful results. The ocean is divided into several layers and integrations are done from interface to interface. For Chebyshev integration, three layers give good results and velocities are required at relatively few depths. Simpson's one-third rule requires equally spaced data and more points, but it is fast on digital computers. Intensities are compared with those calculated using piecewise linear profiles—known to give infinities in intensity calculations.

E10. Ultrasonic Simulation of Shallow-Water Sound Propagation. RICHARD A. RHODES, II AND LYMAN E. GOODNIGHT, III (nonmember), *Florida Presbyterian College, St. Petersburg, Florida.*—A. B. Wood's ultrasonic modeling technique is being used to investigate sound propagation in the shallow ocean in connection with a project on basic aspects of underwater speech and hearing conducted by the University of Florida's Communication Sciences Laboratory. One actual problem simulated is one where measurements were taken in the Gulf of Mexico at the U. S. Navy Mine Defense Laboratory's Stage I, in 31 m of 76°F water over a flat, sandy bottom. The ultrasonic simulation experiment used a water-filled concrete tank 3 ft×4 ft×4 in., side-lined with sound-absorbing rubber matting, and bottom lined with soft rubber to simulate sand. The transducers were two matched pairs of PZT crystals of frequencies 658 kHz and 1009 kHz with wire probes projecting beneath the water surface. Appropriate distance scaling was used to simulate seven frequencies from 0.125 to 8 kHz. Fair agreement was found with the large-scale experiment. Refinements in the simulation technique are being sought. Also calculation by normal mode theory is being undertaken. [Work supported by the U. S. Office of Naval Research.]

E11. Comparative Measurements of Low-Frequency Attenuation in the Deep Ocean Employing Sinusoidal and Explosive Sources. WILLIAM H. THORP (nonmember), *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—This paper describes a deep-water propagation experiment involving the use of both explosive sources and towed sine-wave projectors in the SOFAR channel of the North Atlantic. It was conducted with the objective of narrowing the list of potential causes put forth to explain the well-documented, but unexpectedly large, attenuation coefficients that have been reported by determining whether the observed anomalies stem from fundamental differences in the sources employed or from mechanisms associated with the medium. The study initially considers the conformity of coefficients in the 354–3540-Hz region for the impulsive measurements in this instance with previously reported values. Thereafter results of the present regression analysis are compared with an identical treatment of simultaneous single-frequency data at 1900 and 3800 Hz, and the relative agreement is discussed in terms of the statistical confidence limits of each.

TUESDAY, 14 NOVEMBER 1967 — MEDALLION WEST, 2:00 P.M.

Session F. Psychological and Physiological Acoustics II: Cochlea—Properties and Mechanisms

ERNEST A. PETERSON, *Chairman*

Contributed Papers (10 minutes)

F1. On the Sound-Pressure Transformation from Free Field to Eardrum of Chinchilla. GOTTFRIED VON BISMARCK (nonmember), *Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139* AND RUSSELL R. PFEIFFER, *Washington University, St. Louis, Missouri 63130.*—The sound pressure at the eardrum of anesthetized chinchillas was measured with a probe-tube microphone implanted into the external auditory meatus. The ratio of this pressure to the free-field pressure measured at a point corresponding to the center of the animal's head was determined as a function of frequency (sound pressure transformation function, SPTF). Two distinctly different types of SPTF's were found. They differed by as much as 15 dB in the frequency range 1–8 kHz. Several of the ears that exhibited one type of SPTF showed slight perforations in the tympanic membranes. It appeared that a static pressure difference across the eardrum was responsible for the other type of naturally occurring SPTF's. Both types of SPTF's could be artificially produced by either creating or by equalizing static middle-ear pressure. [This work was carried out at the Massachusetts Institute of Technology and supported in part by the National Institute of Health (Grant), the Joint Services Electronics Program, and NASA Grant.]

F2. Temporary Threshold Shifts in Chinchillas. W. DIXON WARD AND DAVID A. NELSON, *Hearing Research Laboratory, Department of Otolaryngology, University of Minnesota, Minneapolis, Minnesota 55455.*—Detailed audiograms were determined for 20 chinchillas (conditioned avoidance technique). The animals were then exposed for 10 min to successively higher levels (70–110 dB SPL) of a noise specially filtered in an attempt to produce approximately equal values of temporary threshold shift (TTS) at 1, 2, 4, and 8 kc/sec. The observed growth of TTS with level is in essential agreement with results reported by Peters [J. Acoust. Soc. Am. **37**, 831–833 (1965)] on two chinchillas, namely, that they are 10–15 dB more susceptible to TTS from noise than humans. That is, a noise of say 100 dB SPL that produces a certain value of TTS in men will produce the same value of TTS in chinchillas when its level is 85–90 dB SPL. Although group means are quite stable upon replication, within-animal test–retest variability is so great that it is difficult to assign indices of relative susceptibility to the chinchillas. Sources of variability will be discussed. [Research supported by Public Health Service.]

F3. Changes in the Cochlear Microphonic Resulting from Exposure to a 5.0-kHz Tone. G. RICHARD PRICE, *U. S. Army Human Engineering Laboratories, Aberdeen Proving Ground, Maryland 21005.*—As our initial effort in the investigation of the effect of high-intensity sound on hearing, round window potentials were recorded from 34 cats as they were exposed to a 5.0-kHz tone. Exposure times totaled 80 min, and intensities ranged from that intensity necessary to produce a maximum in the cochlear microphonic to 30 dB above that intensity (actual sound-pressure levels ranged from 99 to 138 dB). Measures of the shift in sensitivity following stimulation were made for 12 frequencies at 10 times during the experiment. Data indicate that loss occurs at a rate that is linear in time at high intensities and linear in log time at the lower intensities. The pattern of damage was wide spread with most of the losses occurring below 5.0 kHz.

F4. Hydrodynamic Generation of Distortion in Mechanical Cochlear Models. JUERGEN TONNDORF, *Department of Otolaryngology, Columbia University, New York.*—Displacements of the partition of mechanical cochlear models are invariably larger toward *scala tympani* than towards *scala vestibuli* [J.

Acoust. Soc. Am. **31**, 608 (1959)]. Recent experiments showed that the degree of demodulation of beat signals varies inversely with the viscosity of the cochlear fluids. This finding, together with other previous ones, permits an explanation for the entire phenomenon. For *high* viscosities, the *boundary layers* on both sides of the partition are *broad*. The longitudinal vectors of the two-dimensional trochoidal fluid motion run parallel to the plane of the partition to considerable depths on either side, i.e., the trochoidal pattern is essentially orthogonal. With *lesser* viscosities and *narrowing* boundary layers, however, the horizontal vectors tend to follow the envelopes over the traveling waves, thus causing a "discontinuity" in the trochoidal pattern immediately adjacent to the partition, around the region of maximal amplitude. The notion that this latter "field distortion" might be the direct cause of the observed nonlinearity is supported by two previous findings: (1) the demodulation originates in the region of maximal displacement; and (2) the degree of demodulation varies with the stiffness gradient, i.e., it depends upon the sharpness of the amplitude maximum. [This work was supported by grants from the National Institutes of Health.]

F5. Distribution of Harmonics in the Cochlea. PETER DALLOS AND RICHARD H. SWEETMAN (nonmember), *Auditory Research Laboratory, Northwestern University, Evanston, Illinois.*—While it is now clear that the primary source of harmonics is in the inner ear, there is no agreement on their mode of generation. To study this problem, microphonic potentials (CM) were recorded from guinea pig cochleas with differential electrodes situated in the first and third turns. The distribution of harmonic distortion components in the CM was studied by comparing the relative response magnitudes of the fundamentals, their harmonics, and of tones at the frequency of the harmonics. It is concluded from these experiments that harmonic components in CM are distributed as their fundamentals and not as pure tones corresponding to them in frequency. To study this problem further, we attempted to cancel any one distortion component in the CM with a bone conducted stimulus of the same frequency as the distortion product in question, and of controllable phase and magnitude. While it was a relatively easy task to cancel the CM response for any harmonic component in either turn of the cochlea, we never succeeded in eliminating the distortion in both turns simultaneously. These experiments indicate that internally generated signals (i.e., distortion components) do not generate the same types of traveling wave as do external stimuli. [This research was supported by grants from the National Institutes of Health.]

F6. AC Potentials Inside the Organ of Corti. MERLE LAWRENCE, *Kresge Hearing Research Institute, University of Michigan Medical School, Ann Arbor, Michigan 48104.*—Previous studies have shown the tectorial membrane to be at a nearly zero potential [Ann. Otol. Rhinol. Laryngol. (July 1967)] surrounded by the negative potential of the cortilymph and the positive potential of the endolymph. The first step in determining the role of this polarization in the energy transformation process within the cochlea is a determination of the magnitude of the ac potential in the dc potential regions at various points along the basilar membrane in response to frequencies from 150 Hz to 10 kHz. These recordings are made through electrolyte filled glass microelectrodes with separation of the ac and dc components recorded through the same electrode. Phase shifts as a function of electrode tip position as the various dc potential areas above the basilar membrane are penetrated are also recorded. Results are interpreted as of significance in the process by which vibrations are transformed into the cochlear ac potential. [Research supported by PHS grants.]

F7. Interdependence of the Cochlear Microphonics and Summating Potentials upon the Endocochlear Potential. VICENTE HONRUBIA AND PAUL H. WARD (nonmember), *The Division of Otolaryngology, Vanderbilt University School of Medicine and The Bill Wilkerson Hearing and Speech Center, Nashville, Tennessee.*—The resting potential of the scala media (EP) in the first turn of the guinea pig's cochlea was altered by the application of currents. The EP as well as the cochlear microphonics (CM) and the summating potential (SP) were enhanced when the source electrode was in scala media, whether the sink electrode was in the scala tympany or scala vestibuli. Using the scala media electrode as the sink for the current, caused decreases in these potentials. A linear relationship exists between the changes in the cochlear responses, CM, and SP (in decibels), and the changes in the EP. The SP increased or decreased by 1 dB for a 10% variation in the EP value. The corresponding change in the CM was one-third of the change in the SP. [This work was supported by the National Institute of Neurological Diseases and Blindness Research Grant.]

F8. Prediction of the Volume of the Middle Ear Cavity from Cochlear Microphonic Measurements. FLORENCE E. MCARDLE, *Department of Psychology, Columbia University, New York* AND JUERGEN TONNDORF, *Department of Otolaryngology, Columbia University, New York, New York.*—Previous experiments with a mechanical model of the eardrum and bulla cavity suggested that this system functions in certain respects like a Helmholtz resonator. Subsequent measurements in cats demonstrated an antiresonant peak in the cochlear microphonic sensitivity function that depended upon the dimensions of an experimental opening in the bulla. The present study attempts to use the Helmholtz resonator analogy to predict the volume of the cat's middle-ear cavity directly from its microphonic sensitivity curve. Resonant frequencies (f_{res}) were determined empirically for a series of cats. Using the known dimensions of length l_e and cross-sectional area S of an opening in the bulla, the volume V of the cavity was predicted from $f_{res}=c/2\pi(S/l_eV)^{\frac{1}{2}}$. For each animal, the prediction was checked by filling the spaces with liquid to determine their equivalent volume. [Work supported by grants from the National Institute of Neurological Diseases and Blindness, National Institutes of Health, U. S. Department of Health, Education, and Welfare.]

F9. Auditory Function in Atherosclerotic Rabbits. RICHARD J. VOOTS AND RICHARD LAWRENCE (nonmember), *Department of Otolaryngology and Maxillofacial Surgery, University of Iowa, Iowa City, Iowa.*—Atherosclerosis was induced in a population of rabbits by feeding a high-cholesterol diet for a period of six months. A control group from the same colony was maintanied on an identical diet without the cholesterol. Observations of auditory function were made on five diet animals and four control animals by means of acute round window electrode implants. The action potential response to a standard acoustic click and microphonic sensitivity response curves for gradated 1-sec tone bursts at frequencies from 125 Hz to 15 kHz were recorded from all animals. The AP responses for the diet animals are consistently lower than for the control animals by 10–20 dB. On the other hand, the microphonic responses indicate no obvious differences between the two groups except at 2 kHz, where the diet animals appear to be in the order of 20 dB more sensitive than the controls. The significance of this latter finding, however, is open to question. [This research supported in part by Public Health Service grants.]

F10. Effect of Chloride Deficiency on Cochlear Potentials. TERUZO KONISHI AND ELIZABETH KELSEY (nonmember), *University of Florida College of Medicine, Gainesville, Florida*

32601.—The scala tympani or scala media in guinea pigs was perfused with Ringer–Locke's solution (in the case of the scala media, potassium-rich Ringer's solution) in which chloride was replaced with ferrocyanide, nitrate, glutamate, or acetate. Cochlear microphonics (CM), summating potential (SP), action potential (AP), and endocochlear potential (EP) were recorded from the basal turn before, during, and after the perfusion. Chloride lack in the endolymph did not alter the cochlear potentials, including EP, immediately. Perfusion of the scala tympani with acetate-Ringer's solution caused reversible depression of CM and temporary increase of AP followed by gradual depression. The dc polarization persisted even after chloride ions were removed from the perilymph by introduction of chloride-free solution into the scala tympani. The relationship between the cochlear potentials and the cochlear fluid is discussed. [Work supported by a grant from the National Institute of Neurological Diseases and Blindness, National Institutes of Health, Public Health Service, U. S. Department of Health, Education, and Welfare.]

F11. Development of Hypersensitivity to Acetylcholine by the Denervated Cochlea. E. A. DAIGNEAULT (nonmember), AND R. DON BROWN (nonmember), *Department of Pharmacology, Louisiana State University, New Orleans, Louisiana.*—Previously described experiments [Arch. Int. Pharmacodyn. 162, 20 (1966)] have shown that acetylcholine bromide (20 μg/dose) injected into the arterial supply (i.a.) of the cochlea will produce suppression of the N_1 recorded from the round window of the cat. This depression is similar to that produced by stimulation of the olivocochlear bundle (Nature 192, 1263 (1961)]. With such evidence and other indications from the literature, it is possible to consider that some role may be played by acetylcholine in the cochlea. In an acutely sectioned cat in i. a., injection of acetylcholine is as effective as before the section (25% N_1 depression, 17 observations). We concluded that the system responsive to acetylcholine is in the cochlea proper. Subsequently, cats that had previously had their left eighth cranial nerves sectioned from two to five days were tested as to their responsiveness to i.a. acetylcholine. In six animals, four or five days after section, 2 μg of acetylcholine produced a 22% depression of N_1. This measured hypersensitivity is presented as further evidence that acetylcholine is active in either initiating a mediator release in the organ or is a mediator itself.

TUESDAY, 14 NOVEMBER 1967 — EMPIRE ROOM, 2:00 P.M.

Session G. Engineering Acoustics I: Boundary Layer Acoustics

GERALD J. FRANZ, *Chairman*

Contributed Papers (10 minutes)

G1. Noise from a Simplified Model of a Supersonic Turbulent Shear Layer. H. S. RIBNER, *Institute for Aerospace Studies, University of Toronto, Toronto 5, Canada.*—A layer of two-dimensional square "eddies" separates a main flow U_j from fluid at rest; the eddies move at the mean (supersonic) speed $U_j/2$. The initially plane interfaces will ripple slightly, developing Mach waves to balance internal and external pressures. The Mach wave pressure pattern (noise) is readily calculable. The sound power from unit area of interface comes out to be

$$P=e(\text{peak eddy velocity})^4\,[(U_j/2c)^2-1]^{\frac{1}{2}}/16U_j.$$

In an example (simulated round rocket jet), the effective area is taken as 6π (diam)^2, $U_j/c=6$, peak eddy velocity $=0.1\sqrt{2}\,U_j$. The efficiency (noise power/flow power) comes out to be 0.17%, which is of the order of measured values for rocket noise. [This study was supported by a grant from the Air Force Office of Scientific Research.]

G2. Boundary Layer Noise: The Relationship between Turbulent Shear Stress and the Turbulent Wall Pressure Field. THOMAS BLACK (nonmember), *Mechanical Technology Inc., Latham, New York 12110.*—This paper presents a recently developed model of wall turbulence that provides a new interpretation of the turbulent wall pressure field and hence a new approach to related acoustical and vibrational problems. The theory suggests that the turbulent transfer of momentum is effected by a moving array of horse-shoe vortex structures that are generated and maintained by a spatio-temporal instability of the sublayer flow. As a result of the interaction between these vortex systems and the basic flow, a characteristic pressure signature is generated on the wall directly beneath each vortex structure, the shape and average strength of the signature being related to the local turbulent shear stress distribution. The turbulent wall pressure field is attributed to the motion of the resultant pattern of pressure signatures over the wall, and is seen as an integral and essential part of the turbulent shear stress mechanism. A new "pulse amplitude modulated" model of the pressure field is accordingly proposed in which the pressure at a fixed point on the wall is generated essentially at a single, characteristic frequency (related to local wall shear stress) by the regular or near regular passage of pressure pulses whose amplitudes vary randomly over a range determined by the ratio of boundary layer to sublayer thickness (i.e., over several orders of magnitude).

G3. Radiated Flow Noise. EUGEN J. SKUDRZYK AND G. P. HADDLE, *Ordnance Research Laboratory, The Pennsylvania State University.*—Recently, Lighthill used an improved estimate for the time scale of flow noise, which leads to more than 500 times greater radiation-field pressures for boundary-layer noise. The new theoretical predictions approach the levels generated by rotating cylinders or by large ships. The new theory shows that radiation field is generated in the strongly intermittent outermost part of the boundary layer. The radiation field generated by small vehicles is much greater than that of large ships, because of their smaller stability and the greater intermittency of the turbulence. Radiated flow noise is generated by the boundary-layer turbulence directly. Flow-excited wall vibrations contribute to radiated noise only at frequencies below 500–1000 cycles, provided the boundary layer is thick and the vessel walls thin.

Measurements showed there was practically no difference between the radiation field generated by a relatively thin aluminum-shell buoyant unit, or by one made of solid wood. Radiation-field measurements obtained by three independent methods will be presented: (a) by a hydrophone in the nonturbulent region of the buoyant unit, (b) by a nearfield insensitive hydrophone in the boundary layer, and (c) by three hydrophones at a distance 60 ft from the rising buoyant unit at depths of 250, 150, and 80 ft. [Sponsored by Naval Ordnance Systems Command.]

G4. Effect of Solid Boundaries on Aerodynamic Sound. W. C. MEECHAM, *University of California, Los Angeles, California*.—The theory of aerodynamic sound from turbulence near solid boundaries is re-examined. For nearly incompressible flows, it is shown that the surface integrals, believed to be the source of relatively strong surface sound, may, in fact, merely give the reflecting and diffracting effect of the surface on volume-generated sound. The apparently much larger acoustic effect of hydrodynamic pressure variations may be largely spurious. These characteristics are shown to follow from the use of a simple identity involving the incompressible pressure, which is defined. The present form of the result of this work gives Ribner's source term (known to be equivalent to Lighthill's) for the volume sound. Thus, the total amount of radiated sound can be estimated in the usual way to be proportional to Mach number to the eighth power. There is the usual complication caused by the dependence of the relative turbulence intensity on Mach number. In many boundary-layer applications, the effect of the surface is merely to multiply the intensity of such sound by four. Of course it is true that small objects placed in a uniform flow can generate relatively large velocity fluctuations and cause relatively large amounts of sound to be radiated by the volume sources. It also seems likely that volume sources near small objects will radiate a dipole pattern, though with a Mach number dependence the same as that of volume sources. This is in agreement with experimental observations. [Work was partly done at the Lockheed Palo Alto Research Laboratories; it has been partially supported by ONR].

G5. Acoustic Radiation from Turbulence near a Composite Flexible Boundary. D. G. CRIGHTON (nonmember), *Department of Mathematics, Imperial College, London*.—In an attempt to study diffraction phenomena within the context of aerodynamic noise theory, the problem of radiation from turbulence near an infinite flexible panel is considered, in the case when the panel consists of two half-planes of differing impedances. The Kirchhoff-Powell radiation equations are reduced to a pair of singular integral equations, and a complete formal solution of these equations is given. Various acoustic sources distributed along the line separating the panels are identified and interpreted, and a great simplification of the complete solution is obtained in the high-frequency limit. In this limit, it is shown that, apart from the modification of the turbulent pressure field by a certain distribution of sources along the line of separation, the effect of the composite flexible boundary is merely to reflect rigidly the turbulence-generated sound field. [Research carried out under the Bureau of Ships and General Hydromechanics Research Program, administered by the David Taylor Model Basin, under contract.]

G6. Sound Radiated by an Elastic Plate Excited by Boundary Layer Turbulence.—DAVID FEIT, *Cambridge Acoustical Associates, Inc., Cambridge, Massachusetts 02138*.—The problem of sound radiation by an elastic plate excited by turbulent boundary-layer pressures is investigated. In determining the velocity response, the area of the plate is assumed to be effectively infinite. To provide a more accurate description of the plate motion in the high-frequency range, the Timoshenko–Mindlin thick plate equation of motion is used. Describing the boundary layer pressure field by a narrow-band cross-correlation function proposed by White [J. Acoust. Soc. Am. 40, 1354–1362 (1966)], an expression for the spectral density of the mean-square farfield pressure is obtained. For frequencies above coincidence, the directivity pattern of the mean-square farfield pressure displays two relative maxima. The angles at which the pattern displays its maxima are distinctly different from those that are predicted using classical plate theory. [Work supported by Naval Ship Systems Command, General Hydromechanics Research Program, administered by the Naval Ship Research and Development Center.]

G7. Response of a Transducer System to Turbulent Boundary Layer Pressure Field. G. MAIDANIK AND D. W. JORGENSEN, *Naval Ship Research and Development Center, Washington, D. C. 20007*.—A system of flush-mounted pressure transducers can be considered as a spatial and a spectral filter. The filtering properties of systems consisting of a single transducer, two transducers, and three transducers were previously treated by the authors. [73rd Meeting of the Acoustical Society of America, 21 April 1967, New York, Abstract N/EF8]. This treatment was conducted in $\{\mathbf{x},\omega\}$ space. In this paper, the same problem is analyzed; however, the analysis here is conducted and illustrated in $\{\mathbf{k},\omega\}$ space, the Fourier conjugate of the real physical $\{\mathbf{x},t\}$ space. It is argued that defining the filtering properties of the transducer systems in $\{\mathbf{k},\omega\}$ space offers certain conceptual advantages. This is particularly relevant when one considers the response of the transducer systems to pressure fields that are stationary, both spatially and temporally. The variations in the filtering properties of these systems as functions of the sizes and the sensitivities of the individual transducers and the separations between the centers of the transducers are discussed. It is shown that by properly adjusting the sizes, the sensitivities and the separations, substantial variations in the normalized response of the systems to the pressure field in a turbulent boundary layer can be achieved.

G8. Toward an Acoustically Motivated Length Scale for Noise Environments beneath Disturbed Boundary Layers. E. J. LUKSUS (nonmember), *Lockheed Missiles & Space Company, Sunnyvale, California 94088*.—This paper deals with moderate noise disturbances associated with shock. It is hoped that this work constitutes a bridge between the well-predictable noise due to undisturbed boundary layers and the yet unpredictable noise due to intense shock–boundary layer interactions. Moderately disturbed data is extracted from work published by Speaker and Ailman [NASA CR-486 (1966)], that data being variously scaled to show that the high-frequency behavior is well handled by nondimensionalization according to momentum deficit thickness θ. The prominance of θ substantiates that wall noise originates close to the wall, which makes possible a modification of the displacement thickness δ^*, creating what appears to be another satisfactory acoustic scale. The indication is that boundary-layer acoustic tests should be associated with aerodynamic measurements that emphasize the inner, stress-dominated portions of the boundary layer if information is to be obtained pertinent to the relations between flow disturbances and wall noise.

G9. Turbulent Boundary-Layer Induced Pressures Transduced through a Fluid Loaded Plate. PRITCHARD H. WHITE, *Measurement Analysis Corporation, Los Angeles, California 90025*.—The turbulent boundary-layer pressure field induces flexural vibrations in an adjacent plate, which in turn radiates an acoustic pressure field. The flexural waves in the plate are strongly influenced by the presence of surrounding fluid acting as an additional effective mass or damping. The pressure in the immediate vicinity of the plate is due to the incompressible hydrodynamic nearfield of the subsonic resonantly driven flexural waves. This pressure field decays exponentially away from the plate to reach a limiting value determined by the acoustic components of the turbulence itself. The spatial characteristics of the pressure field also change from the near to farfield, reflecting a downward shift in the wavenumber spectrum. [This work supported by U. S. Navy Ship Research and Development Center.]

G10. Peak Velocity Spectrum of a Flat Plate Excited by Turbulent Flow. WAYNE A. STRAWDERMAN (nonmember), *U. S. Navy Underwater Sound Laboratory, New London, Connecticut,* AND RONALD S. BRAND (nonmember), *University of Connecticut, Storrs, Connecticut.*—A mathematical expression predicting the maximum possible plate velocity spectrum as a function of plate and flow parameters is derived based on results of a theoretical investigation of the response of a simply supported flat plate to excitation by a turbulent boundary later. The maximum possible plate velocity spectrum is termed the *peak velocity spectrum.* Computed *peak velocity spectra* compare favorably with existing experimental information. The resultant expression provides an economical method of predicting the maximum expected plate velocity spectral density for a flat plate excited by a turbulent flow at frequencies above the first few natural modes. Although certain details of the true plate velocity spectrum are not preserved in the plate *peak velocity spectrum,* effects of major parameters are preserved. Thus, the *peak velocity spectrum* supplements the knowledge of flow induced phenomena and provides practical engineering results.

G11. Observation of Turbulence in the Acoustic Boundary Layer. MICHAEL A. COSGROVE, *Acoustical Physics Laboratory, Electrical Engineering Department, University of Rochester, Rochester, New York 14627.*—Turbulence in acoustic flow causes random fluctuations in the particle displacement amplitude. A tracer-particle method has been used to detect boundary-layer turbulence. Micron-size oil drops are injected into an air-filled waveguide, dark-field illuminated, and photomicrographed. With an exposure time slightly longer than one period of oscillation, the image of an oil drop appears on the film as a streak whose length is proportional to the displacement amplitude. Optical reflection from the mirror-smooth boundary produces a companion streak. The streaks are separated by a distance proportional to the height of the oil drop above the boundary. For a 900-Hz progressive wave at a SPL of 140 dB, displacement amplitude measurements agree with laminar-flow theory. At 145 dB, however, measurements are widely scattered about the laminar-flow curve for heights between 70 and 140 μ. It is concluded that the threshold of turbulence at this frequency lies between 140 and 145 dB. [Work supported by the Mechanics Division, AFOSR, Office of Aerospace Research.]

G12. Some Effects of Surface Roughness on Turbulent Boundary-Layer Wall-Pressure Fluctuations in Incompressible Pipe Flow. HOWARD H. SCHLOEMER AND EMILIO C. RECINE (nonmember), *U. S. Navy Sound Laboratory, New London, Connecticut.*—Spectral density, magnitude of the normalized longitudinal and lateral cross-spectral density functions, and convection velocity ratios of wall-pressure fluctuations were measured for smooth and rough surfaces. These measurements were made with small flush-mounted transducers in a 3.5-in.-i.d. pipe air flow facility with centerline velocities ranging from 140 to 238 ft/sec. Two different grades of roughness were used: a coarse, closely packed silica sand grain surface with an average height of 0.088 in., and a finer-grained surface of No. 60 grit commerical sand paper. The effect of increasing grain size was to increase the nondimensional spectral density throughout the frequency band measured and to substantially lower the convection velocity ratio when compared with smooth-wall data. The longitudinal and lateral decay of a particular eddy was more rapid for the rough wall. Experimental evidence indicated that the longitudinal cross-spectral density was more sensitive than the spectral density to surface conditions.

G13. Experimental Study of the Velocity-Dependent Damping of a Turbulence-Induced Vibrating System. MICHAEL E. MCCORMICK (nonmember) AND THOMAS C. RIPLEY (nonmember), *Trinity College, Hartford, Connecticut 06106.*—This paper presents the results of a recent experimental study of the damping characteristics of boundary-layer-induced vibrations of a thin metal ribbon. One side of the metal ribbon was subjected to a turbulent boundary layer in a subsonic wind tunnel. The displacement of the ribbon was measured by a capacitance probe. (The spectral content of the probe's output was analyzed using a wave analyzer.) Each frequency component was then scrutinized to determine its damping characteristics; that is, the -3 dB-down points about each center band/or modal frequency were observed to increase significantly with velocity. This increased damping is attributed primarily to the surface shear stress caused by the fluid on the ribbon. These results can significantly affect the radiated sound from turbulence-induced vibrating surfaces.

TUESDAY, 14 NOVEMBER 1967 — MEDALLION EAST, 2:00 P.M.

Session H. Physical Acoustics II

T. A. LITOVITZ, *Chairman*

Invited Papers (25 minutes)

H1. Acoustic Ion Waves in Gas Plasmas. G. M. SESSLER, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—The generation of acoustic ion waves in gas plasmas is described, and the propagation characteristics of these waves in various ranges of frequency, plasma density, and neutral gas pressure are reviewed. The use of acoustic ion waves in plasma diagnostics is also discussed.

Contributed Papers (10 minutes)

H2. Effect of Dielectric Constant on Relaxation Frequency of $MgSO_4$ Solutions. F. H. FISHER, *University of California, San Diego, Marine Physical Laboratory of the Scripps Institution of Oceanography, San Diego, California 92152.*— Experimental ultrasonic absorption work by Bies (1955) for $MgSO_4$ solutions shows that the primary relaxation frequency is increased to ~200 kHz from 130 kHz as the dielectric constant is lowered from 78.5 to 67 using a water–dioxane mixture as a solvent. This relaxation frequency also increases as a function of concentration, and Fisher (1965) measured a relaxation frequency of around 200 kHz at 0.5 molar concentration. Measurements of the dielectric constant of solutions by Hasted, Ritson, and Collie (1948) indicate that the dielectric constant of a 0.5 molar $MgSO_4$ solution is about 67. These results suggest that the lowering of solvent dielectric constant by adding dioxane is equivalent to the lowering of solution dielectric constant by increasing concentration. The acoustic relaxation frequency data of Bies and Fisher support the hypothesis that the two methods of lowering solution dielectric constant are additive. More accurate measurements should be made to verify previous results. Also, the work of Bies should be extended to lower dielectric constants ($D=45$) where if the additivity assumption holds the relaxation frequency should show an increase to about 200 kHz based upon the 0.5 M work by Fisher for a solvent with $D=56.5$.

H3. Propagation of Shear Ultrasonic Waves below the Relaxation Frequency in Molten Glasses—Highly Sensitive Phase-Lock Detection System. P. B. MACEDO (nonmember) AND J. H. SIMMONS (nonmember), *National Bureau of Standards, Washington, D. C. 20234*.—The pulse operation of the high-temperature ultrasonic interferometer using transmission of shear waves will be described. A feature that made this work possible was the use of a lock-in amplifier in the video circuit, which suppressed the noise level by 25 dB. Sample data will be shown where both the velocity and absorption are independently measured with an absorption per wavelength of 4.9 Np.

H4. Volume Viscosity of Liquid Antimony. J. R. SMIRNOW (nonmember) AND J. JARZYNSKI, *Physics Department, The American University, Washington, D. C.*—Liquid antimony exhibits many of the anomalous properties of water. For example, its density increases on melting, and its compressibility shows a minimum as a function of temperature. As a further study of its properties, the absorption of ultrasound was measured in molten antimony at temperatures up to 900°C. The measured absorption is found to be higher than the absorption due to shear viscosity and thermal conductivity alone. The ratio volume/shear viscosity calculated from the excess absorption is 4.2. The significance of this result is discussed in relation to other properties and the structure of liquid antimony. [Work supported by a grant from the National Science Foundation.]

H5. Effect of Pressure on the Volume Viscosity of Water. C. M. DAVIS, J. JARZYNSKI, AND S. FERMI (nonmember), *Physics Department, The American University, Washington, D. C.*—Measurements of ultrasonic absorption α/f^2 in water have been made over a range of temperatures from 0° to 80°C and at pressures from atmospheric up to 1000 kg/cm². From the measured values of α/f^2, the volume viscosity η_v has been calculated and found to exhibit a maximum versus pressure at all temperatures. [Work was supported by a grant from the Office of Saline Water, Department of the Interior.]

H6. Ultrasonic Attenuation Measurements at Pressures to 10 000 kg/cm². M. PAUL HAGELBERG, *Wittenberg University, Springfield, Ohio 45501*, WALTER H. JOHNSON, JR. (nonmember) AND GERALD HOLTON, *Harvard University, Cambridge, Massachusetts 02138*.—Improvements in previously reported techniques [cf. Bull. Am. Phys. Soc. **77**, 78 (1966); Acoust. Soc. Am. **39**, 1224 (1966)] are described by which reliable ultrasonic attenuation measurements in liquids may be made at pressures to 10 000 kg/cm². Some of the problems involved both in making the measurements and in analyzing the data obtained are described. Results are presented for some typical liquids and the observed absorption compared to the absorption due to shear viscosity. [This work is supported by the U. S. Office of Naval Research.]

H7. Effect of Temperature on the Rotational and Vibrational Relaxation Times of Some Hydrocarbons. GARNETT L. HILL AND THOMAS G. WINTER, *Department of Physics, Oklahoma State University, Stillwater, Oklahoma*.—It has been demonstrated previously that increasing temperature has the effect of increasing the relaxation frequency of a vibrational relaxation and decreasing the relaxation frequency of a rotational relaxation. This effect can be seen clearly in both the ultrasonic absorption and velocity data presented. Measurements covered one to 100 MHz/atm and room temperature to 800°C. In room temperature methane, only the lower part of the rotational relaxation can be seen at 100 MHz/atm, but by 500°C the inflection point of the vibrational dispersion curve has moved up to 1.5 MHz/atm while that of the rotation has moved down to 50 MHz/atm. Two absorption peaks are clearly discernible. By 800°C, the dispersion is continuous over the entire frequency range. Similar measurements on ethylene show roughly the same behavior. In ethane, there are two vibrational relaxations and the effect of temperature on the acoustic behavior is quite different. [Work supported by the U. S. Army Research Office, Durham.]

H8. Molecular Mechanisms Contributing to Ultrasonic Absorption by Tissues. PETER D. EDMONDS, RICHARD Y. C. WANG (nonmember), MATTHEW HUSSEY (nonmember), AND RONALD W. BROWER (nonmember), *Department of Biomedical Engineering and Chemistry, University of Pennsylvania, Philadelphia, Pennsylvania 19104*.—The interpretation of data presented by ultrasonic visualization techniques may be expected to benefit from an understanding of the mechanisms of the predominant contributions to the total ultrasonic absorption coefficients of tissues. Since aqueous solutions of hemoglobin have been most extensively investigated, analyses of specific molecular mechanisms have been undertaken with relation to the structure of this macromolecule and its environment. The contributions that could arise from reactions occurring at the charged side chains of residues of arginine, lysine, aspartic acid, and glutamic acid were considered since these residues occupy positions with highest frequency of occurrence at the hydrophilic exterior of the hemoglobin molecule. On the assumption of a structural mechanism, a family of possible single relaxation curves was calculated from which one member could be selected with knowledge of an experimental value of relaxation time. We conclude that basic side chains make significant contributions, which are larger than those of acidic side chains. [Work supported by a grant from the National Institutes of General Medical Sciences.]

H9. Sound Absorption and Velocity in CO_2/D_2O Mixtures. F. DOUGLAS SHIELDS, *University of Mississippi, University, Mississippi*.—Sound absorption and velocity measurements have been made in CO_2/D_2O mixtures at 300, 400, and 500°K. The results show that D_2O is from one-third to one-fifth as efficient as H_2O in de-exciting the bending mode vibration of CO_2 in a molecular collision. The collision efficiency of D_2O for de-exciting CO_2 decreases with temperature at an even greater rate than does that of H_2O. The large difference between D_2O and H_2O invalidates the chemical affinity explanation of effect of H_2O and CO_2. An approximate semiclassical calculation of the transition probability indicates that it is possible to attribute the relaxation to an interchange between the vibration of the CO_2 and the rotation of the H_2O or D_2O.

H10. Vibrational Relaxation in Nitrogen with Water Vapor and Carbon Dioxide Contaminants. MALCOLM C. HENDERSON, RAYMOND COAKLEY (nonmember), OSFS (nonmember), AND JACQUES BRY (nonmember), *The Catholic University of America, Washington, D. C., 20017*.—The pressurized Kundt's tube developed in this laboratory has been used to measure the napier (relaxation) frequency of nitrogen in both binary and ternary mixtures with water vapor and carbon dioxide.

Carried out at about 175°C, where the relaxing vibrational specific heat of nitrogen becomes large enough to be readily measureable, both contaminants shift the Napier frequency upward by amounts simply proportional to their molar concentrations: water vapor by 300 Hz per mole%, carbon dioxide by 250 Hz per mole%. However, preliminary work indicates that when both are present the shift is not the simple sum of the individual effects, but that there is considerable synergism. Such a synergistic effect is indeed to be expected on theoretical grounds. [This work was supported by the Office of Naval Research.]

H11. Measurement of Viscoelastic Properties of Polymers by Ultrasonic Methods. J. R. Smithson, *United States Naval Academy, Annapolis, Maryland.*—Yasaku Wada [J. Phys. Soc. Japan 15, 2324 (1960)] has proposed a method of calculating the real and imaginary parts of the bulk modulus of a polymer from measurements of the sound velocity and absorption coefficient of a water suspension of the polymer. An analysis of the propagation of measurement errors through the bulk modulus calculation, shows that errors in sound velocity are magnified in the calculation, but existing methods of velocity measurement are sufficiently accurate to allow worthwhile results. A feasibility study of the method has been made using some special suspensions of polybutadiene which were prepared by Goodrich Rubber. These suspensions were unique in that they contained very narrow bandwidths of particle sizes. Studies have considered particle size, concentration, temperature, sound frequency, and effect of dissolved salts and soaps in the suspensions. Results indicate that the method has a considerable range of usefulness. [Supported by the Office of Naval Research.]

H12. Determination of Attenuation Constants for Air in Tubes by Standing Wave Measurements. W. L. Beech (nonmember) and J. V. Sanders, *Naval Postgraduate School, Monterey, California 93940.*—Attenuation constants were precisely measured for air, at ambient conditions, contained in a rigid-walled tube. Standing waves were excited by a piston with the fundamental having a frequency of 93.8 Hz and a bandwidth of 1.12 Hz. Attenuation constants were determined both by the conventional technique of measuring the frequency difference between the half-power points on the resonance curve (Method A), and by measuring the microphone and accelerometer output voltages at resonance (Method B). The experimental results to the 40th harmonic were compared with the Rayleigh–Kirchoff theory that predicts a $f^{\frac{1}{2}}$ dependence. Both methods yield a frequency dependence of $f^{0.54\pm0.3}$. The attenuation at the fundamental frequency was $(1.03\pm0.03)\times10^{-2}$ m^{-1} by Method A and $(1.10\pm0.04)\times100^{2}$ m^{-1} by Method B, approximately 10% and 15% higher than the theoretical value for a smooth tube without end losses. Method B which does not require accurate frequency determinations, has not been previously reported.

Tuesday, 14 November 1967 — Silver Chimes East, 2:00 p.m.

Session I. Speech I: Sound-Spectrum Analysis; Synthesis

John C. Lilly, *Chairman*

Contributed Papers (10 minutes)

I1. Spectrographic Analyses of English Consonants. Yukio Takefuta, *The Ohio State University, Columbus, Ohio.*—The purpose of this study was (1) to analyze the English consonants spectrographically and (2) to calculate the relative contribution of each acoustic correlate to the distinction of one articulatory feature from another. Three male speakers of General American dialect recorded 24 English consonants five times in intervocalic positions of five different vowels, /a, e, i, o, u/. Three types of spectrographic display were made for all consonants: sonagram, section, and amplitude display. Quantitative measures were made for each of three aspects of acoustic analysis (intensity, duration, and frequency spectrum). The acoustic features were interpreted in relation to the articulatory features of manner or place of articulation. Each correlate was also considered as a possible distinctive feature or an acoustic signal of the consonants, and the magnitude of this signal was a measure of its strength. The index of signal detectability was calculated to quantify the relative contribution of the acoustic features.

I2. Real-Time Colored Display of Sound Spectrogram. Yasuaki Nakano (nonmember), Takeshi Nakayama (nonmember), Tsuneji Koshikawa, and Tanetoshi Miura, *Central Research Laboratory, Hitachi Ltd., Kokubunji, Tokyo, Japan.*—If signal spectrograms are displayed in real time, so-called brightness modulation display is used in general. Because of the limited dynamic range of brightness modulation of CRT, details of the spectrum are lost very often. In order to overcome such defects and be able to observe instantaneous spectrum patterns more clearly and easily, we constructed new display apparatus in which effective use of visual function is considered. In this system, instantaneous spectrum amplitude is quantized in 11 levels with the step of 4 dB, and each amplitude level is converted to predetermined hue resembled to contour map. Furthermore, these colored instantaneous spectrograms are memorized in sky-sign like fashion in a certain time interval. As the results of displaying various acoustic signals, it is confirmed that color display of this type is more useful than monochromatic display for the visual discrimination of the characteristics corresponding to each signal differences.

I3. Spectrum Analysis of Various Asian Dialects. S. Joseph Campanella, *Communications Satellite Corporation, Washington, D. C.* and R. C. Kloffenstein (nonmember), *Melpar, Inc., Falls Church, Virginia.*—It was the objective of the study reported here to analyze a variety of Asian dialects for the purpose of discovering spectrum distribution differences that may be correlated to the dialects. The dialects subjected to the analysis were North (10), Central (10), and South (10) Vietnamese; Cantonese (5), Mandarin (5), and Fukienese (4) Chinese; Cambodian (15) and the highland tribal dialects of Jari (3), Bahnar (3), Sedang (3), Rade (3), Cham (3), Koho (3), and Hre (11). The number of individuals sampled in each dialect is shown by the digit in the parentheses. Confidence in the results should be rated accordingly. All samples were taken from male talkers. Spectrum analysis was performed by means of a 31-channel 1/3-oct band filter bank over the frequency range from 20 to 20 000 Hz, scanned at a rate of 60 Hz. The speech source data were collected on high-quality portable tape recorders and these, after appropriate editing to remove unwanted background, were played into the 1/3-oct band spectrum analyzer. The spectra were then digitized and transferred to an IBM 1410 computer for analysis for mean spectrum distribution for each dialect. Voiced and unvoiced events were segmented into separate groups and separate spectra for each of these classes thereby obtained. For the summary analysis, all spectrum scans of the voiced type and unvoiced type were individually averaged to produce an average voiced event spectrum and an average unvoiced event spectrum for each dialect. A tally of the number of spectra for each excitation class for each dialect also permitted the estimation of the ratio of voiced to unvoiced excitation that is characteristic of each dialect. The summary analysis showed that, generally, the voiced event mean spectrum differed only in minor detail as a function of dialect. The same was found true of the unvoiced event mean spectrum. The most variable parameter appeared to be the percentage of time occupied by voiced excitation. This parameter varied from 76% for Mandarin-Chinese to 90% for Cambodian, Jari, and Hre. [Dr. Campanella was employed at Melpar, Inc. at the time of performance of the study reported in this paper. The work was performed under Contract Booze–Allen Corp. subcontract 5-66-1]

I4. Coarticulation and the Locus Theory. PIERRE DELATTRE, *University of California, Santa Barbara, California.*—The effects of coarticulation on formant transitions do not ". . . necessitate a re-evaluation of the locus theory. . ." [S. E. G. Ohman, J. Acoust. Soc. Am. **39**, 151 (1966)]. In fact, no acoustic data on human speech behavior could be used to alter this theory since it is essentially based, not on the production of human speech, but on the *perception* of controlled acoustic variables by human brains. The locus concept simply describes the direction and extent that the second- and third-formant transitions of a consonant must take *in order to contribute to the perception* of place of articulation. It does not claim that the specifications of formant transitions defined by the locus are observed by all human speakers under all conditions. Earth speakers are only human. But it does assume that when, in human speech, a transition does not point to the appropriate locus, that transition *does not contribute to the perception* of the appropriate place of articulation. Tables of VCV coarticulation will be examined to find how the perception of place of articulation is possible when the interference of coarticulation prevents a transition from pointing to the appropriate locus. It will be shown that in every such case, other cues substitute for the locus cue.

I5. Pseudospeech Synthesis and the Phonette. H. M. TRUBY, *Division of Newborn Infant and Phonetic Studies, Communication Research Institute, Coconut Grove, Florida.*—Direct programming intimacy, spanning 17 consecutive years, with, respectively, the Haskins Labs PB-2 and Octopus, the Swedish Royal Institute of Technology Ove I, LEA, Lucia, progressively developmental stages of Ove II, etc., and the IBM San Jose Terminal Analog Synthesizer, (both Mark I: manual patterns, and Mark II: computer-stored, diphone-nucleus nuclei) has led the author of this paper to consider strongly the term "speech" as applied to the products of electromechanical "synthesis" operations. The linguistics (sic) implications of the contrivance "pseudospeech" to these products seem highly apt to this investigator, who at one time proposed the acronym TAPS for the IBM device (and thus: Terminal Analog Pseudospeech Synthesizer) and presented a paper before the Xth Annual National Conference on Linguistics entitled, "A Phonetic Macro-matrix for Pseudo-speech Synthesis." The relevance of this iconoclastic, but perhaps useful, consideration to questions of listener judgment(s) and other psychological-testing procedure(s) is discussed, both in its own light and in light of a variable segmental unit, the *phonette*, proposed as a part of the Acta Radiol Logica Monograph *Acoustico-Cineradiographic Analysis Considerations* (Truby, 1959), a 227-page publication including 1100 sound spectrograms of American-English word utterances, each with correlated oscillogram, over-all-intensity record, pitch curve, and duplex oscillogram. Synthesis and speech-transmission analysis have much to offer future communication research, provided they are not delimited by misdefinition and equivocal evaluation criteria.

I6. Digital Formant Synthesizer for Speech Synthesis Studies. LAWRENCE R. RABINER, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Recent work on speech synthesis by rule has led to the development and evalation of a novel digital formant synthesizer. The synthesizer was simulated on a computer at a 20-kHz sampling frequency. Among the novel points are an excitation source for the unvoiced component of voiced fricatives; a voicebar for the stop gap period of voiced stops; a novel source filter providing the voiced excitation for the synthesizer and the elimination of the higher pole correction network for voiced sounds by exploiting the properties of sampled data systems. A comparison (in terms of the frequency response for vowels) between digital and analog synthesizers will be made. A brief discussion of the advantages and disadvantages of cascade and parallel synthesizers will be presented. Examples of synthetic speech will be presented.

I7. Control Rule of Vocal-Tract Configuration. AKIRA ICHIKAWA (nonmember), YASUAKI NAKANO (nonmember), AND KAZUO NAKATA (nonmember), *Central Research Laboratory, Hitachi Ltd., Tokyo, Japan.*—Objectives of the research is to find out a simple but an effective rule for the control of vocal-tract configuration in the synthesis of speech. One of the following two alternatives is often selected as the control rule: (1) independent linear interpolation, (2) mathematical descriptions with several parameters. The former has an advantage of simplicity but also has suffered from over simplified deformation. The latter is hardly applicable to consonant. The proposed processes are as follows: (1) Define three key points for each target configuration; they are maximum area points of front and back cavity V_1, V_2, and maximal constriction point P between these two. (2) Trace the trajectory of these key points between two target configurations in a certain time fashion. (3) Calculate the temporal change of each section by the interpolation from starting target value to ending target value with considering a restriction due to the pass of V_1, V_2, and P point. The proposed rule has two advantages: (1) applicable to any target configurations including consonants, (2) the restrictions avoid the results from over-simplified deformations. Some of the result will be shown with aids of our vocal tract pattern display.

I8. Vocal-Tract Perturbation Functions. JOHN M. HEINZ, *Department of Otolaryngology, The Johns Hopkins University School of Medicine, Baltimore, Maryland.*—In recent years, Webster's horn equation has been widely used as the basis for an approximate description of the acoustic behavior of the human vocal tract. Starting from an assumption of the validity of this equation, a procedure for determining for an arbitrary vocal-tract configuration those first-order changes in the configuration that will effect changes in one and only one formant frequency (or, alternatively, one and only one lip impedance pole or zero) is presented. Examples illustrating the iterative use of this procedure for determining vocal-tract configurations that correspond to specified impedance pole-zero patterns are given together with a discussion of some of the practical difficulties that arise in this application. [This work was supported in part by Special Fellowship from the National Institute of Neurological Diseases and Blindness, Public Health Service while the author was a guest at the Department of Speech Communication, The Royal Institute of Technology, Stockholm, Sweden.]

I9. Control Rule of Voice Source to Improve the Naturalness of Synthetic Speech. TAKESHI NAKAYAMA (nonmember), AKIRA ICHIKAWA (nonmember), AND KAZUO NAKATA (nonmember), *Central Research Laboratory, Hitachi, Ltd., Tokyo,* *Japan.*—Though it is rather difficult to define the naturalness of speech clearly separated from its articulation, we assume we can roughly do so. Naturalness, in this sense, still means many aspects of speech. Some of them are very important in the practical application of synthetic speech, since it is very hard to grasp a whole meaning of synthetic speech quickly, if the speech was not provided with good naturalness. The objectives of the research is to find out the way to improve the naturalness of synthetic speech, through the control of its voice source characteristics. *Firstly*, analysis has been made on the voice source characteristics, especially on pitch frequency and intensity envelop of glottal waves. *Secondly*, rules has been derived from the data with several assumptions for the simplification. *Thirdly*, synthesis was made according to the rules and listening test was performed. The results tell us that some aspects of naturalness, which are mostly derived from syntax of the sentence, are improved by the relatively simple rule, but emotional state and voice quality are not so easily communicated to the listeners by synthetic speech. Further details of the analysis, synthesis, and results of listening test will be presented at the meeting.

I10. Acoustic Implications of Interspecies Communication. JOHN C. LILLY, HENRY M. TRUBY, ALICE M. MILLER (nonmember), AND FRANK GRISSMAN (nonmember), *Communication Research Institute, Coconut Grove, Florida.*—When, in interspecies exchange of information, the dominant modes of the communication are vocal and acoustic, some of the physical limitations imposed upon the communication are shown by the hearing curves and the acoustic energy output curves. Additional limitations are shown in the difference frequency limen curves and in certain time-pattern perception limitations. One such communication system is the interlock or feedback between a single individual *Homo sapiens* (Hs) and a single individual *Tursiops truncatus* (Tt). The speech output of Hs was measured with a high-frequency microphone; detectable amounts of energy were found above the generally accepted speech band, up to the order of 60 kc. This high-frequency energy is detectable by Tt as is shown by its hearing curve from 400 Hz to 160 kHz. The usually accepted limits for Hs–Hs transmissions for 100% intelligibility are from approximately 100 Hz–10 kHz. The Hs-Hs feedback control thus is limited to approximately 10 kHz; i.e., energy above this limit is not controlled nor used by Hs. In the Hs–Tt transmission, this energy functions as an adequate stimulus to Tt. Thus, to Hs this high-frequency energy is unknown noise but can be mistaken by Tt for signals. Further consideration of the physical limitations on Hs-Tt vocal exchanges are considered with a feedback diagram and quantitative calculations based on the hearing curves and the sonic–ultrasonic output of Hs and Tt. Thus, in treating another species with a different hearing curve and a different sonic energy output, it is imperative to have quantitative physical measures in order to find channels that are adequately open both ways between the different species. Such considerations determine necessary acoustical and electrical transforms of the outputs and inputs from and to each member of the different species. [Work supported in part by AFOSR and NIH, NINDB.]

I11. Cryprint Categories in Newborn-Infant Cry-Sound Analysis. H. M. TRUBY, *Division of Newborn Infant and Phonetic Studies, Communication Research Institute, Coconut Grove, Florida.*—The first 10 yr of an ongoing, long-range, acoustic and acoustigraphic (sound-spectrographic, jet-written oscillographic, and other visible-acoustic) analysis program have been centered upon the investigation of paranatal, immediately postnatal, and neopostnatal cry-sound in human infants. This program has recently been extended to include the examination of postnatal periods beyond the drastically transitional neonatal period. Of the thousands of sound spectrograms of newborn-infant cry-sound examined, at no time has any sequence "repeated," either in the same infant or in different infants, yet, based upon 20 yr of intimate experience with sound-spectrographic analysis of adult speech and other acoustic phenomena, it seems evident to the author of this paper that each infant unambiguously demonstrates his individuality in his cry-sound patterning... i.e., his cryprint. The prospect of ultimately validating cryprinting as a reliable analog to footprinting and fingerprinting was proposed at 60th ASA Meeting in San Francisco, 1960 as was the prospect of establishing the visual-acoustic patterning of neonatal, or even paranatal, cry-sound as supplementary clinical-diagnosis criteria in the early detection of brain damage and other serious neurological impairment. The research-in-progress status of these studies will be discussed and demonstrated in acoustigraphic and acoustic form. [Work supported by NIH: NICHD, and by United Cerebral Palsy Research and Educational Foundation.]

I12. The "Voiceprint" Mystique. RALPH VANDERSLICE AND PETER LADEFOGED, *University of California, Los Angeles, California.*—Proponents of the use of so-called "voiceprints" for identifying criminals have succeeded in hoodwinking the press, the public, and the law with claims of infallibility that have never been supported by valid scientific tests. The reported experiments comprised *matching from sample*—subjects compared test "voiceprints" (segments from wideband spectrograms) with examples known to include the same speaker—whereas law enforcement cases entail *absolute judgment* of whether a known and unknown voice are the same or different. There is no evidence that anyone can do this. Our experiments, with only three subjects, show that: (1) the same speaker, saying the same words on different occasions without any attempt to disguise his vioce, may produce spectrograms which differ greatly; (2) different speakers, whose voices do not even sound alike, may produce substantially identical spectrograms. As with phrenology a century earlier, "voiceprint" proponents dismiss all counter evidence as irrelevant because the experiments have not been done by initiates. But our data show that the use of "voiceprints" for criminal identification may lead to serious miscarriages of justice.

TUESDAY, 14 NOVEMBER 1967 BURGUNDY EAST, 2:00 P.M.

Session J. Underwater Acoustics II: Signal Processing

CHARLES J. LODA, *Chairman*

Contributed Papers (10 minutes)

J1. Comparison of Sonar Equation and Decision Theory Approaches to Passive Detection. PHILIP L. STOCKLIN, *Raytheon, Submarine Signal Division, Portsmouth, Rhode Island.*—The sonar equation for passive detection, usually expressed in decibel form, equates recognition differential (N_{RD}) to target, propagation and sonar system power measures and is the basis for recognition differential (N_{RD}) figure of merit and signal excess performance prediction methods. If the problem of detection performance prediction is formulated using decision theory, detectability index d is the major performance measure. Comparison of the two approaches yields (a) a quantitative measure of N_{RD} in terms of d, bandwidth, processing time, and type of processing; (b) lack of a finite optimum bandwidth for detection, in general, (c) inherent difficulty of the decibel approach in finding theoretical detection performance over a very broad bandwidth, (d) means and the extent to which detection performance prediction can be factored with operational requirements versus design factors.

J2. Space–Time Structure of Likelihood Ratio Sonar Receiving Systems. James F. Bartram, *Raytheon, Submarine Signal Division, Portsmouth, Rhode Island.*—The conventional way of designing a SONAR receiving system is to subdivide the design task into at least two major parts: the array and the receiver proper (the transmitter of an active SONAR system is beyond the scope of this paper). The array processes in space; that is, it combines suitable delayed space samples of the acoustical field, resulting in a voltage output that is a function of time. The receiver proper processes in time; that is, it makes use of the input time function over a finite observation time, yielding at the end the decision of the required type. The present paper investigates the factors involved in the synthesis of a best signal processing system spanning the whole domain from ocean to decision. Having formed a suitable space–time sampling plan, one can write the requisite likelihood ratio, which prescribes the best signal processing operations. In general, these operations will consist of inextricably mixed space and time processing. The question then suggests itself: What are the necessary and sufficient conditions for separation of the likelihood ratio system into space processing followed by time processing, or vice versa? This subject is treated in this paper.

J3. Optimum Threshold Detection with Adaptive Beamforming in a Class of Non-Gaussian Signal and Noise Fields. David Middleton, *Raytheon, Submarine Signal Division, Portsmouth, Rhode Island.*—The threshold (i.e., weak input signal) structures for the optimum binary detection of general signals in a broad class of non-Gaussian noise fields are determined for a variety of situations that include both adaptive and preformed beams. The nonnormal noise fields represent various types of process with slowly fluctuating intensities, e.g., "fading". They are derived from the multivariate X^2-gauss distribution and include as important special cases the multivariante and multiple Lapace densities. For the important case of independent sampling of the background noise (multiple Laplace density), the optimum threshold structure requires in various orders: ideal limiting ("superclipping"), squaring, beamforming, and integration of the input. Several examples involving independent space and/or time sampling of the input signals are operationally diagrammed, and the general approach is shown to yield DIMUS structures as special cases. Superclipping occurs in all instances, basically because of the non-Gaussian falloff of the tails of the amplitude distribution of the noise.

J4. Matched-Filter Processing of Hydrophone Arrays. J. L. Flanagan, L. Landgraf (nonmember), and D. J. MacLean (nonmember), *Bell Telephone Laboratories, Incorporated, Murray Hill, New Jersey.*—Multipath transmission between a sound source and a simple beamforming array degrades the gain of the array. Matched-filter processing is one means for combating multipath distortion and for maintaining array gain (Parvulescu). In addition, matched-filter processing produces a volume focusing of the source (Schroeder). That is, where the simple beam former locates the source only in bearing, the matched filter bounds the source in three dimensions. Simple relations for array gain and for received signal-to-noise ratios are derived for the matched filter. Undersea signals are recorded from five hydrophones for source ranges out to 100 miles. The signals are digitalized and processed by transversal matched filters simulated on an IBM 7094 computer. Measurements are made of array gain, and rough estimates are made of the focal volume size. Calculated and measured performance of the matched-filter array are in satisfactory agreement and suggest potential improvement in detection capability over the simple beam former. The improvement in gain is found to be on the order of the number of significant multipaths.

J5. Optimum Transmit and Receive Beam Patterns for Sonars Operating Against Reverberation. W. S. Liggett, Jr. and D. A. Pitassi (nonmember), *Raytheon, Submarine Signal Division, Portsmouth, Rhode Island.*—Since the shape of the transmit beam pattern affects not only the strength of the echo but also the spacial correlation of the reverberation, optimum design of this pattern may improve performance. Following transmission of a narrow-band pulse, our system decides if an echo from a slowly moving target is present in the volume reverberation and noise. The target direction and range are known *a priori*. Our transmit pattern may be adjusted by applying amplitude weightings and phase shifts to the transducer inputs. Using a well-known reverberation model, we obtain the beam former for the Bayes optimum receiver. Once the optimum receiver for arbitrary transmit pattern has been found, further reduction in the error probabilities is obtained by the proper choice of transmit shading coefficients. As examples, we investigate line arrays with half-wavelength spacing. When the reverberation power is much greater than the noise power, the optimum transmit and receive patterns do not have their maxima in the target direction. When noise that is independent between hydrophones dominates, unshaded patterns are optimum.

J6. Effect of Spatially Uncorrelated Noise on the Maximum Array Gain Processor. Benjamin F. Cron and Clair J. Becker (nonmember), *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—Several authors have shown that large array gains may be obtained by weighting individual elements with the optimum amplitude and phase factors. It is known that spatially uncorrelated noise, α, such as electronic noise, will decrease this maximum array gain. The purpose of this study is to obtain a quantitative value for this loss. Vertical, linear, equally spaced arrays from two to 10 elements are analyzed for both an isotropic and a surface generated noise field. Calculations on each array with elements spaced from one eighth to one wavelength have been made showing: (1) array gain as a function of the direction of "look" for several values of α, (2) array gain versus α for various spacing intervals, and (3) array gain versus the number of elements for given values of α. The loss of array gain increases as the number of elements increases for a given α. However, for the worst case of 10 elements spaced one-eighth wavelength apart, α could be 20 dB below the ambient noise level and still give much better results than conventional processing.

J7. On Measuring the Stationarity of Ambient Sea Noise. B. E. Parkins (nonmember) and B. P. Bogert, *Bell Telephone Laboratories Inc., Whippany, New Jersey*—Efforts to determine the low-frequency spectrum of ambient sea noise have shown that there exist slow variations in the noise power in the ocean. These variations amount to a modulation of a "short-term" estimate of the frequency spectrum of the noise. A search for this modulation has been made by filtering the sea noise from a single deep hydrophone into nonoverlapping frequency bands and then detecting and recording the envelopes of the variations within these bands. The envelope variation has been investigated by examining the self- and cross-spectra of the set of envelopes for the bands analyzed.

From these results, the coherencies of the covariation between bands of frequency versus the modulation periods were calculated. For a stationary random process, the coherency should vanish. The data analysis, thus far, which includes stretches up to one day in length, shows very small coherence between bands centered an octave apart. This implies that either the noise is almost stationary or the "modulation" is incoherent for these frequency separations. [Work supported by the U. S. Navy Office of Naval Research.]

J8. Value of Memory in Sequential Detection. R. A. ROBERTS AND C. T. MULLIS (nonmember), *Department of Electrical Engineering, The University of Colorado, Boulder, Colorado 80302.*—In detection models, one usually assumes the detection device can retain any function to any degree of precision necessary to optimize the performance of the detector. Under some conditions, it may be necessary to limit the soft memory of the detector for reasons of size, expense, etc. The problem of optimizing the detection performance of a sequential detector with a limited soft memory is investigated. For the sequential detection problem under study, the optimum full memory detector must store in soft memory two variables—one deterministic and one random. The random variable L gives the cause probabilities at each stage of observation. The limited memory detector is permitted only an m-bit register to store the random variable L. Limited numerical results indicate that a 3-bit register gives performance close to the full memory detector. [This work supported by NSF grant.]

J9. Estimation of Periodic Signals in Noise. THOMAS G. KINCAID, *General Electric Company, Heavy Military Electronics Department, Syracuse, New York* AND HENRY J. SCUDDER, III (nonmember), *General Electric Company, Research and Development Center, Schenectady, New York.*—Two methods for the estimation of a periodic signal in additive noise are presented. Both assume a finite time sample of the signal plus noise is available with a large signal-to-noise ratio. Both minimize the mean-square error between the estimate and the sample. In the first method, estimates for the period of the signal and the complex amplitudes of its harmonics are derived. In the second method, estimates for the period of the signal and the waveform of one period are derived. Assuming white noise, the expected values and variances of the period estimates are derived. The estimates for the period are found to be the same by both methods. The estimate is unbiased and has a variance inversely proportional to both the signal-to-noise ratio and the cube of the number of periods in the sample. The expected values of the waveform estimates are derived, and the estimates are found to be biased. [This work was supported in part by the U. S. Office of Naval Research.]

J10. Algorithmic Selection of Acoustic Waveforms. NORMAN W. LORD, *Hudson Laboratories of Columbia University, Dobbs Ferry, New York 10522.*—The choice of acoustic waveform is studied within the category of widely spaced FM pulses for the goal of optimum resolution between overlapping arrivals. For a given average frequency and pulse time length, it is analytically apparent that there exists a wide variety of monotonic F-t shapes that have the same average frequency, time length or number of cycles, and frequency difference between initial and final cycles. It is then shown how to program the calculation of autocovariance among all such waveforms using a preliminary calculation of a subset of the partitions of an integer sum representing the pulse time length. The number of integers in the partition is restricted to be the number of cycles with the smallest integer representing the highest frequency and the largest representing the lowest. The most prevalent choice, for an optimum decline among the autocovariance side-lobe peaks, has a frequency variation that is small at the ends of the pulse and high near the center. It is also possible to seek along the courses of spreading the frequency extremes and changing the pulse length for F-t shapes that further minimize the side lobes. [Hudson Labs. of Columbia Univ. Informal Documentation No. 137. This work was supported by the U. S. Office of Naval Research.]

J11. Measurement of the Spectral Density of Ocean Reverberation. ARTHUR S. WESTNEAT (nonmember), *Raytheon, Submarine Signal Division, Portsmouth, Rhode Island.*—An experiment is described that provides measured estimates of the spectral density of reverberation under typical ocean conditions with a spectral dynamic range in excess of 30 dB. Measurements were made with a precisely controlled narrowband pulse at 15 kHz on a platform moving at various speeds between 0 and 15 knots, both across and with the wind, and in deep as well as shallow water under two sea conditions. A 10° array was employed that was alternately pointed at 90° and 0° relative to ships heading. Tape recordings of the reverberation echoes were analyzed through digital comb filters, a digital Fourier transform, and a fast Fourier transform program, to verify observed results and to obtain the widest available dynamic range compatible with cost of reduction. Tape flutter spectral components, which were found to obscure the reverberation spectra, were removed by controlled digital sampling. The effects of range on spectral width are examined by sliding the sampling window through the echo period. Comparisons are made of spectra with calculated beam pattern effects, and with assumptions of a normal distribution of scatterer velocities.

J12. Acoustic Spectrum Analysis with Coherent Optics. GEORGE E. STANFORD (nonmember), *U. S. Navy Underwater Sound Laboratory, Bermuda Research Detachment, FPO New York 09560.*—An optical spectrum analyzer has been assembled at USL to investigate the application of one- and two-dimensional transforms in SONAR signal processing. In the one-dimensional mode, the analyzer was used to examine a sample of low-frequency marine noise. Of special interest was the ability to view the dynamic spectrum of the noise field. Some parts of the spectrum fluctuated rapidly, whereas other portions varied very slowly. The stronger lines were more nearly stationary, varying slowly in intensity and spectral position. Some of the lines split into numerous less intense lines, eventually regrouping to form a single well-defined spectral line. The analyzer was used also in the two-dimensional mode, performing as a combined direction finder-spectrum analyzer. A synthetic signal, simulating the output from a continuous linear array, increased the intensity of the two-dimensional spectrum by a factor of 8.7 compared with that of the one-dimensional, single-channel spectrum. The two-dimensional transform also produced information regarding signal directionality, appearing as a rotation of the spectrum in the output plane.

J13. Analysis of Doppler Shift of Sonar Backscatter. ROBERT HARTLEY (nonmember), *Raytheon Marine Research Department.*—A digital procedure for the analysis of Doppler shift of SONAR backscatter in presented along with examples from 4.5 and 14 kHz sonar signals. The reduced data are compared with the results of other researchers and existing theory. The digital program is run on a CDC 8090 with 4096 words of memory. The program makes use of a single sideband algorithm for heterodyning the digital data into the range of a fixed band of simulated digital resonant filters. Outputs of the simulated filters are typed in symbolic form with each symbol representing an increment of 3 dB. There are 12 possible symbols giving a system dynamic range of 36 dB. The method employed eliminates the effects of wow and flutter from the original analog recordings and produces a variable width spectrum about the transmit frequency.

TUESDAY, 14 NOVEMBER 1967 MONTE CARLO HOTEL, 2:30 P.M.

Monte Carlo Session. Ultrasonic Visualization II: Medical Specialities

(*Joint session* AIUM/ASA)

DENNIS WHITE AND WILLIAM MCKINNEY, *Joint Session Chairmen*

Invited Paper (25 minutes)

MC1. Ultrasonic Encephalography. CHARLES GROSSMAN (nonmember), *Department of Neurology, University of Pittsburgh.*—A review of clinical applications of diagnostic ultrasound, in conjunction with EEG, will be presented.

MC2. Ultrasonic Stereotaxis. DOUGLAS GORDON, *Richmond, Surrey, England.*—Ultrasonic visualization of human tissues has aimed at obtaining information simultaneously from a number of different depths from the surface. In consequence, only weakly focused transducers have been employed. In stereotaxic surgery when the highest possible accuracy is essential in three dimensions, it is preferable to sacrifice all other considerations to reducing the size of the focus to the minimum and to move the transducer bodily so that the focus scans a single plane of the body which can be in any of the three anatomical axes. As echoes are obtained from what is virtually a point source, the effect of the angulation of the surface is much less important than when plane waves are used. Using conventional gating circuits, it is possible to select only those echo pulses that occur exactly at the focus. These are recorded automatically on electrosensitive paper, either directly or with enlarged scale, by use of a pantograph. The problems involved in obtaining short well-damped pulses for diagnosis and high acoustic power continuously for destructive purposes from the same transducer have been overcome by employing high harmonics for diagnosis and a spherically ground ceramic bowl with no added damping. In actual laboratory experiments on cats, it has been possible to localize the major blood vessels to 0.1 mm three dimensionally and to distinguish between the surface of the hemisphere and the roof of the lateral ventricle only 5 mm below.

MC3. Ultrasonography in Ophthalmology. GILBERT BAUM (nonmember), *Albert Einstein College of Medicine, Bronx, New York 10461.*—This paper is a historical review of the use of ultrasound in ophthalmology. The paper will review its earliest therapeutic applications, the use of A and B Mode for diagnostic purposes, the localization of intraocular and intraocal foreign bodies by both the A and B Mode techniques, ultrasonic measurement of eyesize, and the use of intense focused ultrasound for the production of focal chorioretinal lesions. The present state of these applications will also be discussed.

MC4. Ultrasound Cardiogram in Clinical and Physiological Studies. CLAUDE JOYNER (nonmember), RICHARD PYLE (nonmember), AND JOHN GRUBER (nonmember), *Edward B. Robinette, Foundation and Cardiovascular Clinical Research Center Hospital of the University of Pennsylvania, Philadelphia, Pennsylvania.*—Ultrasound cardiograms have been obtained from over 3000 subjects over the past 6 yr. The accuracy of this method for the assessment of mitral valve disease, as described in earlier reports, has been confirmed in this large patient study. This method of external, safe study has been found to equal cardiac catherization in accuracy when judged by findings at operation. Equally reliable evaluation of tricuspid valve disease has been obtained, and actually found superior to catherization in preoperative assessment. Valve substance, pliability, and mobility can be determined from the ultrasound records. The prediction of valve characteristics determining whether replacement of a valve with a prosthesis is needed, has been quite accurate. This preoperative information is not obtained from catherization studies. The behavior of the mitral valve, accounting for the opening snap in mitral stenosis, and the Austin–Flint murmur of aortic regurgitation, have been defined by simultaneous record of direct video display of the mitral valve ultrasound with the phonocardiogram. The use of the mitral ultrasound as a reference for analysis of sound records will be presented. The ultrasound method of determining pericardial fluid has been confirmed as valid but subject to record error and misinterpretation. The ultrasound cardiogram is 100% accurate in prediction of mitral valve sizes and valve leaflet character. The accuracy in determination of pericardial fluid is less reliable.

MC5. Sonographic Interpretation—Pitfalls and Breakthroughs. LAJOS I. VON MICSKY (nonmember), *Department of Obstetrics and Gynecology, College of Physicians and Surgeons, Columbia University and Bioacoustical Laboratory, Woman's Hospital, St. Luke's Hospital Center, New York, New York.*—Contrary to the impression conveyed by recent publications, the problems arising in diagnosite ultrasonics are not primarily of a technical nature. The difficulty now lies in deciphering the vast amount of information recorded on the sonograms and in elucidating the tissue's structural characteristics responsible for the echo pattern obtained. Since relatively little is known and understood of the physical mechanisms attending the propagation of high-frequency acoustic waves in biological materials, the first step toward intelligent interpretation is to explore new theoretical and experimental avenues for gaining insight into the sources of tissue echoes. The use of morphologic interpretive criteria alone has been rendered grossly inadequate by the well-known fact that a number of different conditions can produce sonographic patterns that appear nearly identical on visual inspection. The various methods of data processing practiced in our laboratory are discussed in detail, emphasizing a general tendency twoard some form of quantification. New techniques involving color-translating isodensitracing, ultrasonic holography, quantitative evaluation of compound scan sonograms, and automatic pattern recognition are mentioned.

MC6. Diagnostic Application of Pulse Echo Ultrasound to the Abdomen. JOSEPH H. HOLMES (nonmember), *University of Colorado Medical Center, Denver, Colorado 8022α.*—Present studies indicate that ultrasonic pulse echo compound scanning techniques are of value in demonstrating intra-abdominal pathology and thus may prove to be a valuable diagnostic aid in a wide variety of diseases. Equipment used employs either contact scanning or water-path scanning. The liver, kidney, and spleen all transmit sound well and thus can be readily outlined to determine size and position. In the liver, which transmits sound well, abnormal echo patterns are observed with such lesions as cirrhosis, tumor, abscess, chloecystitis, congestion, and hepatitis. Renal cysts and tumor give abnormal echo patterns. The ultrasonic technique has had greatest diagnostic application on examining abdominal masses. Fluid filled structures like the stomach and the bladder are outlined readily by ultrasound, and it is possible to delineate structural distortion, pressure from adjacent structures, and to estimate the amount of fluid present. The pancreatic area, abdominal aorta, and vena cava can be visualized in some patients with appropriate use of time-varied gain and depth control. Ultrasonic controls that must be used to obtain proper visualization will be discussed.

TUESDAY, 14 NOVEMBER 1967 — SILVER CHIMES EAST, 8:00 P.M.

Workshop Session. Underwater Acoustics Workshop: Ships and Sonar

DONALD ROSS, *Chairman*

Invited Papers (20 minutes)

Underwater acoustics is closely tied to the ships that take it to sea. On the one hand, the sonar system is affected by the ship (surface and submarine), and the equipment design and performance reflects the ship characteristics. On the other hand, the installation of the sonar system aboard the ship influences the ship's performance and its design. It is the purpose of this workshop to discuss these sonar system ship interactions.

WU1. Effects of Sonar Domes on Ship Performance in Smooth Water. G. STUNTZ, *Naval Ship Research and Development Center, Carderock, Maryland.*

WU2. Motion Performance Characteristics of ASW Ships. RAY WERMTER (nonmember), *Naval Ship Research and Development Center, Carderock, Maryland.*

WU3. Hydromechanics Problems of Variable Depth Sonar. REESE FOLB (nonmember), *Naval Ship Research and Development Center, Carderock, Maryland.*

WU4. Effect of Ship Motions on the Design of Sonar Equipment. G. W. BRADSHAW, *Naval Undersea Warfare Center, San Diego, California.*

WU5. Influence of Acoustics on the Evolution of the Submarine. WALTER L. CLEARWATERS, *USN Underwater Sound Laboratory, New London, Connecticut.*

WEDNESDAY 15 NOVEMBER 1967 — SILVER CHIMES EAST, 9:00 A.M.

Session K. Ultrasonic Visualization III: Image Formation, Conversion, and Display

WESLEY L. NYBORG, *Chairman*

Invited Papers (25 minutes)

K1. Acousto Optics and Their Application to Ultrasonic Visualization. H. W. JONES (nonmember), *University of Wales, Swansea, England.*—This contribution reviews the present position of acousto-optics in relation to ultrasonic visualization. The basic principles relating to matching, reflection, and refraction are briefly reviewed. Detailed consideration is given to the problem of matching at the front face of an image convertor. The attenuation in liquids and solids is considered and related to the diffraction effects that arise in lenses and mirrors. The results of calculation of the relative importance of diffraction and geometric aberrations in lenses and mirrors accounting for absorption and mode conversion effects are given. The use of zone plates and other artifices to improve performance is discussed. Finally, the problems of ultrasonic illumination are considered in relation to reflection and transmission modes of operation.

K2. Investigation on Electronic Image Conversion in Pulsed Operation. R. POHLMAN, G. RADIG, AND E. SCHÄTZER, *Laboratorium fur Ultraschall, Aachen, Germany.*—The two possible methods of the electronic ultrasonic image conversion—the CW or impulse-reflex technique—are confronted and compared. For short distances, an unequivocal superiority of the impulse-reflex-technique results. The low coverage at high frequencies proves to be hindering, which mainly depends on the damping in the transmission channel and by this on the frequency, but furthermore, also on the shape of the object that is to be imaged. At equal number of image elements, the diameter of the plate must increase at lower frequencies. Therefore, two possibilities to increase the mechanical solidity were investigated. By this it follows that at equal solidity, a shell construction allows a diameter that is larger by factor of 20. The materials usable for such a construction were tested concerning their application in the ultrasonic image conversion technique, and results are reported. The second part deals with the application of the impulse-reflex technique. The charge image on the transducer plate produced by an impulse can be scanned only partly during the short period of an impulse. Therefore, the application of a sampling method is necessary. The electronic arrangement for this method is described. The limits of the technique are specified.

K3. Sensitivity Predictions for Ultrasonic Image Tubes. WILLIAM R. TURNER, *Vitro Laboratories, Silver Spring, Maryland 20910.*—The sensitivity of the electronically scanned ultrasonic image tube is assessed in terms of its potential development as well as the existing state of the art. The piezoelectric conversion element is treated not only as an open-circuit potential source, but also as an energy converter with storage capability. Low- and high-conductance modes of transferring the electronic image signal to the output circuit through secondary emission are contrasted, and the effective use of output coupling networks described for the high-conductance mode. Noise sources are identified and evaluated as to their relative importance in current tube designs, and also as to the steps that might be taken to reduce noise significantly. By a summary chart relating the equivalent noise intensity to the video bandwidth, published sensitivities for existing tubes are compared and referenced to potential sensitivities that might be achieved during further image tube development.

Contributed Papers (12 minutes)

K4. Real-Time Display of Sound Holograms by KD*P Modulation of a Coherent Light Source. H. BOUTIN AND R. K. MUELLER, *Bendix Research Laboratories, Southfield, Michigan.*—It is possible to reconstruct optically a sound hologram by reflecting a laser beam from the pattern formed on the water surface [R. K. Mueller and N. K. Sheridon, Appl. Phys. Letters **3**, 328 (1966)]. The acoustically illuminated object is then viewed in real time but the quality of the reconstruction is poor because of aberrations due to streaming and oscillatory surface disturbances. A way of improving the real time reconstruction process is presented, in which the holographic pattern is recorded on the surface of a piezoelectric crystal (quartz), immersed in the water, and then forms the end wall of a Sokolof-type electron tube. A scanning electron beam reads off the corresponding voltage pattern obtained on the surface of the piezoelectric crystal. It produces an output signal that is then applied to a deuterated KDP crystal (KD*P) scanned by an electron beam synchronized with that of the first tube. It is therefore possible to modulate the index of refraction of KD*P (Pockels effect) and obtain phase modulation corresponding point by point to the holographic pattern recorded on the quartz surface. Illumination (reflection or transmission) of the crystal with laser light will permit optical reconstruction of acoustically illuminated objects. This device permits a scale change of the hologram that reduces aberrations.

K5. Reference Waves in Synthesized Acoustical Holograms. A. F. METHERELL, *Douglas Advanced Research Laboratories, Huntington Beach, California 92646,* AND H. M. A. EL-SUM, *Atheron, California 94025.*—The reference wave necessary for recording a hologram of an object irradiated with acoustical waves A. Metherell and H. El-Sum, Appl. Phys. Letters (July 1967)] can be provided externally by appropriately controlling the phase between two coherent signals, the one that derives the sound source, and the other which is used as a reference by summing it electronically with the signal diffracted from the object. The shape of the wavefront of the reference wave and the angle it makes with the object wave are determined by the phase shift between the two waves on the recording plane. This in turn depends on the wavelength, the recording technique (continuous or sampled [A. Metherell, H. El-Sum, J. Dreher, and L. Larmore, Appl. Phys. Letters **10**, 277 (1967); J. Acoust. Soc. Am. (Oct. 1967), the type of the sought hologram (narrow or wide angle, Fourier, Fresnel, etc.), and the final optical reconstruction. These and other factors affecting the choice and control of the reference beam will be discussed. Experimental results of acoustical holograms taken with various sound frequencies (9–21 kHz) will be presented.

K6. Face Plates for Image Converters. HENRY B. KARPLUS, *IIT Research Institute, Chicago, Illinois.*—Electronic image converters have been built largely as demountable systems with attached vacuum pumps for experimental purposes. Permanently sealed systems with large face plates requiring a high vacuum on the inside and good acoustic coupling to the transducer through the wall have had only limited success. The approach used here has been a separation of the vessel wall and transducer material. Three bonding systems were investigated: (1) cement, silicone rubber showed good stability; (2) low melting point alloys, and (3) direct heat bond of barium titanate to an Inconel surface.

K7. Nearly Spherical Lenses. H. R. FELDMAN AND G. E. LORD, *Applied Physics Laboratory, University of Washington, Seattle, Washington.*—A unidirectional geometrically perfectly focusing lens can be made by shaping appropriately the interface between homogeneous lens media. For good geometric focusing over a wide angle, the lens should be as spherical as possible. A single component lens that focuses a perfect image of an object at infinity is a conic section of eccentricity $1/n$, where n is the index of refraction of the lens material. The limitation on n due to the lack of transmissive materials with very low acoustic velocity requires that such a lens be quite aspherical. This difficulty is not inherent in multicomponent lenses. Perfectly focusing shapes have been determined for two-component lenses with spherical outer surface. The meridional and sagittal focusing properties of these lenses have been studied for obliquely situated objects. With a proper choice of lens parameters, the internal interface is not required to deviate from sphericity by more than about 2%, so that the attainment of a perfect geometric focus in the forward direction does not seriously degrade the imaging of objects situated at very wide angles.

K8. Acoustic Field in a Luneberg Lens. G. E. LORD AND H. R. FELDMAN, *Applied Physics Laboratory, University of Washington, Seattle, Washington.*—A Luneberg lens, which is a perfectly focusing omnidirectional spherical lens, has been subjected to a wave analysis. Plane radiation incident on a fluid lens was assumed. The separability of the scalar Helmholtz equation for the lens region permits the representation of the internal acoustic field as an expansion in Legendre polynomials and radial functions. The radial equation obtained from the separation was solved in integral form from which the necessary recurrence relations were obtained. Beam patterns at and relatively near the focal point were obtained for moderate wave number-aperture products ($k_0 \sim a80$). Calculations for high frequencies are hampered by the lack of sufficiently general schemes for function generation. The beam patterns so obtained are quite similar to those obtained for comparable spherical fluid lenses of uniform refractive index. Lens gains and beamwidths do not differ significantly in the two cases. In general, the side lobes for the Luneberg lens are relatively high and decrease relatively slowly in magnitude as the polar angle increases.

K9. Basic Problems in Underwater Acoustic Imaging. J. BLAINE DAVIDSON, *Florida Atlantic University, Boca Raton, Florida 33432*, AND JOSEPH HULL, *Litton Industries, Electron Tube Division, San Carlos, California 94070.*—The basic elements of an underwater acoustic imaging system are presented. Each element that introduces serious problems in the design and construction of a practical system is discussed and known ways of solving the problems are outlined. Problems of specular reflection, range-definition tradeoffs, size, weight, cost, lenses, deep submergence pressure, and data rate are presented. Techniques utilized in the HYDROCON system are used as illustrations of one practical embodiment of the principles of a system. Finally, new ideas are presented to suggest areas in which techniques now under development may provide solutions to some of our current problems.

WEDNESDAY, 15 NOVEMBER 1967 — EMPIRE ROOM, 9:00 A.M.

Session L. Architectural Acoustics

CYRIL M. HARRIS, *Chairman*

Contributed Papers (10 minutes)

L1. Control of Air Distribution Noise in Critical Auditoriums. R. M. HOOVER, *Bolt Beranek and Newman Inc., Cambridge, Massachusetts 02138.*—To achieve low background noise levels in critical air-conditioned auditoriums, such as concert halls, it is essential to control the noise generated by air flow through air outlets and in terminal distribution ducts. In this paper, the air distribution systems of several auditoriums, and the corresponding background noise levels, are examined to help define appropriate noise-control measures for the design of silent air distribution systems. In addition to the evaluation of several existing air distribution systems, laboratory noise data on various types of air outlets are presented and related to field results.

L2. Acoustical Design and Testing of the Centenary Auditorium of Madras University, Madras. B. S. RAMAKRISHNA, *Indian Institute of Science, Bangalcre, India.*—The paper describes the acoustical design and testing of a 3250-seat multipurpose auditorium. Among the problems discussed here is the question of what reverberation time is optimal for a hall that uses a sound reinforcement system to augment the direct sound. The design and performance of a sound reinforcement system for the hall is also described. The hall has a volume of about 800 000-ft^3 and a midfrequency RT of 2.3 sec when empty and 1.7 sec when full. The bass ratio is approximately 1.3. The maximum variation of the sound level over the seating area is 4.5 dB. The noise level (on *C*-weighting) in the fully occupied hall is 42 ± 1 dB without air conditioning and 52 ± 1 dB with air conditioning.

L3. Acceptability of Footfall Sounds in Apartments. B. G. WATTERS, *Bolt Beranek and Newman Inc., Cambridge, Massachusetts 02138.*—We have studied the acceptability of intruding footfall sounds in apartments. In controlled laboratory experiments, the signal frequency range, signal duration, and ambient noise levels have been varied to correspond to variations in floor surface compliances, room RT's, and other factors typical of real apartments. The test subjects, experienced apartment dwellers, compared the acceptability of footfalls with intruding speech at that just-intelligible level previously found marginally acceptable in buildings. The footfalls they judged as acceptable as the reference speech signal were just detectable, i.e., very close to the masked threshold. The condition of constant acceptability, as footfall level, center frequency, bandwidth, and reverberation are varied, requires the quantity $(WT)^{\frac{1}{2}}S_0/N_0$ to be held constant, where W is the effective signal bandwidth, T is the effective signal duration, S_0 and N_0 are the signal and noise intensities per hertz. A proposed floor rating scheme is shown to be similar to one developed for the Insulation Board Institute. However, neither of these compares with the ISO R140 procedure that deals, not with footfalls, but with a more intense impact. This work was supported by the Armstrong Cork Company.

L4. Comparison of Three Methods of Rating Floors for Impact Noise. T. MARINER AND G. R. SPALDING, *Armstrong Cork Company, Lancaster, Pennsylvania.*—About *200* floors were examined in the laboratory using as sources the ISO tapping machine and female walkers wearing hard high heels and prejudged to be consistent vigorous walkers. Data were reduced to impact noise rating (INR), for tapping machine noise [see T. J. Schultz, J. Acoust. Soc. Am. **36**, 729 (1964)]; and, for walking noise, to loudness [see T. Mariner and H. W. W. Hehmann, J. Acoust. Soc. Am. **41**, 206 (1967)], and to a detectability rating [see B. G. Watters, paper L3]. While loudness and detectability rating for these tests correlate generally within the estimated experimental error ($\pm10\%$ loudness) involved in the walking tests, there are larger discrepancies for some of the louder floors that subjective judgment had classed as "more detectable than loud" before detectability had been calculated. We conclude (1) that detectability rating is more significant than loudness rating and (2) that the INR is inadequately correlated with suppression of walking noise, whether related to loudness of footfall noise or to detectability of footfall noise.

L5. Statistical Analysis of Sound Absorption by Audience. K. V. GEETHA (nonmember) AND B. S. RAMAKRISHNA, *Indian Institute of Science, Bangalore, India.*—It is proposed that the sound absorption coefficients of acoustic materials, in general, be considered as random variables since their values depend upon the sound fields in which they are measured. This viewpoint is particularly appropriate for audience absorption since clothing, seating, etc. vary randomly. On this assumption, it is possible to decide between the classical hypothesis that the audience absorption is proportional to their number and Beranek's hypothesis that audience absorption is proportional to the occupied area, by analyzing the variance. A two-way classification of the total absorption of a number of halls is made with seating density and seating area as the row and column variables, respectively. The absorption of the hall in the ith row and jth column is written as $a_{ij}=\mu+r_i+c_j+\epsilon_{ij}$, where μ is the mean of all entries, and r_i and c_j are the row and column effects, respectively. The ϵ_{ij}'s represent contributions from other sources and interactions. Tests for the hypotheses $r_i=0$ and $c_j=0$ indicate that audience absorption varies significantly with seating area but not with seating density, supporting Beranek's hypothesis.

L6. Application of Random Walk to Reverberant Room Acoustics. DAVID LUBMAN, *LTV Research Center, Western Division, Anaheim, California 92803.*—In a reverberant room driven with a steady high-frequency tone, it has been reported that values of squared sound pressure are distributed through space according to the chi-square distribution of two dimensions. Under what circumstances is this distribution to be expected, and what is the physical significance of two statistical dimensions? We gain insights, and some answers, by casting the problem into the form of a random walk in two (sic) dimensions, wherein the reverberant field is viewed as the sum of interfering, travelling waves of the dirving-tone frequency. The chi-square distribution is shown to be a limiting case for many waves under broad conditions. With few waves, the problem is more complex. Useful and revealing results are found by examining special cases for which the orderly transition to the limit can be observed.

L7. Improved Rotating Diffuser for Reverberation Chambers. EDWARD D. LAWLER, *York Division, Borg-Warner Corporation, York, Pennsylvania 17405.*—An experiment was conducted with a rotating diffuser consisting of several acoustically reflective panels in the form of a truncated cone to determine if it would give more efficient diffusion of a reverberant room's modal patterns than a typical flat-panel rotating vane. The diffuser construction was such that deviations from the purely conical shape could be made to determine the effect of changes in shape on the degree of diffusion. Since the amount of diffusion in most low-frequency regions varies with the location of the rotating vane within the room, it was also considered to be possible that a spatial traverse of the vane would result in a space-averaged diffusion and more uniform over-all modal response. To determine the effect of such a spatial traverse, the drive mechanism of the conical diffuser was modified so that the vane's axis of rotation simultaneously traversed an asynchronous orbit. It was found that the orbital-rotating conical diffuser gave a significant flattening of the room's low-frequency modal response and was much superior to the rotating flat diffuser.

L8. Impedance Tube Used as a Source of Known Acoustic Power. WILLIAM A. JACK, *Johns Manville Research & Engineering Center, Manville, New Jersey.*—In an open-ended tube with negligible losses, the standing wave pattern is probed for maximum and minimum SPL's at the time the tube is generating a pure-tone steady-state level in the room under test. The power emitted is a function of these levels and the tube-section area. The room average SPL is a function of this power and the effective sabins in the room. These sabins can be evaluated. If more absorbing material is introduced, the effective sabins added can be evaluated. With simple sources, such as a fixed rpm fan or a fixed voice coil current loudspeaker, possibly valid objections have been made that adding more absorbing material could change the source output in an unknown manner. With the tube method, it is of no consequence whether or not the changed acoustical environment reacts to change the power, as the output is measured for each condition. It would be desirable to extend the method to handle warble tones and narrow bands of random noise.

L9. Improved Automatic Decay-Rate Meter. A. J. PRESTI, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—The original design of the automatic decay-rate meter [J. Acoust. Soc. Am. 40, 1267 (A) (1966)] has enabled us to measure the sound decay rate in a room easily and successfully, and has also enabled us to gain more insight into the operation of this device. This paper will review some of the design concepts and introduce an improved version designed to further reduce setup time and ambiguities.

L10. Sound Decay Rates in Various Conference Rooms as Measured by the Automatic Decay-Rate Meter. F. K. HARVEY AND A. J. PRESTI, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—The automatic decay-rate meter described by A. J. Presti [J. Acoust. Soc. Am. 40, 1267 (A) (1966)], has been used to examine the sound decay in a number of conference rooms, with special emphasis on those having fast decay rates. Results are compared with determinations from several different types of oscilloscope presentations. Advantages and limitations in the use of the decay-rate meter will be discussed.

L11. Sabine Absorption Coefficients of Aluminized Rubber. RICHARD W. CASE, EUGENE W. VAHLKAMP, AND LAWRENCE E. KINSLER, *Naval Postgraduate School, Monterey, California.*—Since 1930, the work of Eyring and others has indicated that the effective Sabine absorption coefficient of highly absorbing materials is not given by α_i, the randomly incident energy absorption coefficient, but is instead given by $-\ln(1-\alpha_i)$. This indicates that for $\alpha_i>0.63$, the Sabine coefficient should be greater than 1.0. A search of the literature provides many measured values of 0.99 for the Sabine coefficient but rather surprisingly only one or two measured values in excess of 1.0. This paper reports on reverberation measurements made at frequencies from 10 to 100 kc/sec in water contained in a small tank completely lined with either flat or cone structured aluminized butyl rubber. The maximum value measured for the Sabine coefficient was 2.56 which corresponds to $\alpha_i=0.923$. In carrying out these experiments, decay rates in excess of 25 000 dB/sec were measured through use of a Memoscope. Information as to the advantages of using a Memoscope in making reverberation measurements will be presented.

L12. Excessive Reverberation Time for Organs? DARIEL FITZROY, *San Rafael.*—Whether or not organists and organ builders are demanding excessively reverberant environment is posed as a problem for consideration. The point is brought up that too long reverberation periods could result in an undesirable coloring of what is heard.

WEDNESDAY, 15 NOVEMBER MEDALLION WEST, 9:00 A.M.

Session M. Engineering Acoustics I: Radiation and Scattering

K. JEROME DIERCKS, *Chairman*

Contributed Papers (10 minutes)

M1. Acoustic Scattering of FM Pulses from Aluminum Spheres in Water. E. E. MIKESKA, *Defense Research Laboratory, The University of Texas, Austin, Texas 78712* AND R. HICKLING, *General Motors Research Laboratories, Warren, Michigan.*—Experimental measurements of acoustic backscattering of FM pulses from a family of precision-solid and hollow-aluminum spheres of 7 in. o.d. are presented and compared with computed echo pulse forms. The present paper is an extension of work reported by Diercks, Goldsberry, and Hickling for 5-in.-diam spheres [J. Acoust. Soc. Am. 40, 1258 (A) (1966)]. The spheres studied had i.d./o.d. ratios of 0.95, 0.8, and 0.0, with either air or water inside the hollow spheres. The incident FM pulses had a nondimensionalized center frequency of $ka=20$ (approximately 54 kHz) with nondimensionalized frequency ranges of 15–25 and 10–30 during the pulse. Incident pulse lengths were selected in the nondimensionalized form $\Delta\tau=0.5$, 1.0, or 5.0, where $\Delta\tau$ relates the range extent of the pulse to the radius of the sphere according to $2\Delta\tau=c2\Delta t/a$, where $2\Delta t$ is the pulse length in seconds, c is the velocity of sound in water, and a is the radius of the sphere. Detailed comparison of experimental and theoretical results shows generally good agreement in frequency structure and amplitude for FM pulses having a range extent equal to or shorter than the sphere diameter. Comparisons for longer pulses are also shown. [This work was done under contract with the U. S. Office of Naval Research, Code 463, and the U. S. Naval Ordnance Test Station, Pasadena, California.]

M2. Acoustic Scattering by a Soft Cylinder. T. G. GOLDSBERRY AND O. D. GRACE, *Defense Research Laboratory, The University of Texas, Austin, Texas 78712.*—Backscattering coefficients have been computed for an infinite, soft cylinder over the range of acoustic radii, *ka*, from 0.5 to 25. Scattering patterns in a plane perpendicular to the cylinder axis have been obtained for selected *ka* values. Experimental measurements of acoustic scattering in water were obtained using a cylinder 10 cm in diameter machined from low-density Styrofoam. The experimental measurements cover the *ka* range 1.2 to 21. Excellent agreement between theory and experiment were obtained over this *ka* range and indicate both the validity of the computations and the usefulness of low-density Styrofoam as an acoustically soft material in underwater acoustics. [This work was done under contract with the Office of Naval Research, Code 463.]

M3. Causality in Sound Scattering from Elastic Cylinders. H. ÜBERALL, *The Catholic University of America, Washington, D. C.* AND P. UGINČIUS, *U. S. Naval Weapons Laboratory.*—We have investigated the causality aspects of a plane acoustical wave being incident on an elastic cylinder, and producing circumferential waves on the cylinder surface. The phase and group velocities of the latter were obtained by performing a Watson transformation on the normal mode series and searching for the zeroes of the denominator in the complex plane. An incident wave packet was constructed by a Laplace transform; then the speeds of the head of the wave C_i can be found by closing the contour to the right. This procedure also furnishes the paths of propagation of the pulse during the scattering process: One finds that the waves enter and leave the cylinder surface towards the outside at critical angles ϑ^i given by $\sin\vartheta^i=c/C_i$, towards the inside at $\vartheta_{l,t}^i$ given by $\sin\vartheta^i=c_{l,t}/C_i$, where c is the sound speed in the surrounding medium, and $c_{l,t}$ are the longitudinal and transverse speeds. Two types of circumferential waves are obtained; Rayleigh type, with $c_i>c$, and Franz type, with $C_i\cong c$.

M4. Scattering by Coated Prolate Spheroids. C. YEH,* *Electrical Engineering Department, University of Southern California, Los Angeles, California 90007.*—Most previous theoretical analyses [See W. P. Mason, Ed., *Physical Acoustics* (Academic Press Inc., New York, 1964)] were carried out with the assumption that the scattering object was either an elastic sphere, an elastic cylinder, or a rigid spheroid. Because of the importance of the penetrable spheroid problem with regard to the scattering by nonspherical bodies, the task of solving this problem was carried out recently [presented at the 73rd Meeting of the Acoustical Society of America in New York during April 1967 by C. Yeh]. The purpose of the present presentation is to consider the problem of the scattering by a coated prolate spheroid. A rigid prolate spheroid coated with a sheath of acoustic material characterized by (ρ_1,c_1) is assumed to be immersed in a acoustic medium characterized by (ρ_0,c_0). ρ and c are, respectively, the density and the sound speed of the medium. Emphasis will be placed upon the presentation and interpretation of the numerical results in the resonance region. [Work supported by the Office of Naval Research.]

* Starting 1 Sept. 1967, C. Yeh will be with the Dept. of Engineering, Univ. of California, Los Angles, Calif. 90024.

M5. Luneburg Lens and Wave Theory. C. ALLAN BOYLES, *Ordnance Research Laboratory, The Pennsylvania State University, State College, Pennsylvania, 16801.*—The wave theory for an underwater Luneburg lens is discussed in terms of a generalized wave equation, which differs from the ordinary wave equation in that it contains a term involving the gradient of the material density of the lens. We then consider the special case where, to a first approximation, the density of the lens has a constant value equal to that of water. This case is of some importance since it corresponds to the compliant tubing lens built and tested by W. J. Toulis [J. Acoust. Soc. Am. **35**, 286 (1963)]. For this special case, we determine the analytic solution of the wave equation when the lens is being irradiated by plane waves, and then present the results of a numerical evaluation of that solution. In particular, we calculate the acoustic pressure field along the axis of the lens and the pressure receiving patterns. These results give us the gain and beamwidths for the range $1\leq k_0a\leq 30$ (k_0 the wavenumber in water, a the radius of the lens).

M6. Time-Domain Analysis of Broad-Band Refraction and Diffraction. H. A. WRIGHT, *Space Systems Division, Avco Corporation, Lowell, Massachusetts 01851.*—The refraction and diffraction of short-duration transients is analyzed by means of the Kirchhoff integral. A solution for acoustic refraction is obtained which expresses the disturbance on the axis of a spherical liquid lens (diffraction plate) in terms of: the lens (plate) parameters, the lens (plate)-to-observation point distance, and the source-emitted disturbance. It is shown that at distances small compared to the image distance, the disturbance on axis will be predominantly that emitted by the source; that the disturbance in the vicinity of the image point will be functionally the derivative of the source-emitted disturbance; and that at distances large compared to the image distance, the disturbance will be predominantly the inverse of that emitted by the source. Experimentally observed pressure-time records resulting from refraction and diffraction of short-duration electrodeless spark pulses are described that confirm the analytical results.

M7. Diffraction by a Wedge. JOHN D. INGRAM, *Rice University, Houston, Texas 77001.*—The diffraction of waves by a wedge-shaped boundary separating two media has long been an open question in acoustics. In this paper, an exact solution is presented for such a physical system with arbitrary wedge angle and source placement. Kontorovich–Lebedev transforms are used to solve the equations of motion and to analyze the boundary conditions. This analysis results in a singular integral equation for the transformed potential. Function theoretic methods are used to reduce this equation to a Fredholm equation with a symmetric kernel. The exact solution is then given in terms of the eigenfunctions of the symmetric kernel. The principal contributions of this work are the production of a joint distribution for the radial eigenfunctions (with two distinct wave speeds) in cylindrical coordinates and the method used to reduce a nondegenerate singular integral to a Hilbert–Schmidt equation. These two techniques, however, not only allow the solution of the scalar problem reported here, but also show the way to solution of the vector problems for electromagnetic and elastic fields.

M8. Experimental Investigation of Surface Wave Phenomena by Sound Pulse Techniques. MARY L. HARBOLD AND BRUCE N. STEINBERG, *Department of Physics, Temple University, Philadelphia, Pennsylvania.*—Experimental pulse techniques have been developed to verify directly the existence of *creeping* or *circumferential* waves that arise from theoretical diffraction theory as developed mathematically by the Watson transformation acoustic step pulses and sine-burst pulses of varying duration and frequency, produced by a transducer using modulation of an ion cloud, are scattered from a rigid sphere in air. The detection transducers, positioned near to and imbedded in the surface of the scatterer, yield measurements of both the amplitude attenuation and velocity of the *creeping* wave recorded by high-speed, multitrace oscilloscope techniques in conjunction with *sample/hold* digital data storage. Electronic gating circuits switch the transducers *on* and *off* individually or in coincidence as the pulse front advance. The transducers are synchronized in tandem with the pretrigger, trigger, and delayed trigger circuitry of six independent oscilloscope traces. [This work was supported by the U. S. Navy, Office of Naval Research.]

M9. Simultaneous Variation of Frequency and Angle in Acoustic Studies. W. M. BARSS, *University of Victoria, Victoria, British Columbia.*—The effect of submerged objects on a beam of sound in water is strongly dependent on both the frequency of the sound and the angle of incidence at the interface. Instead of measuring the intensity of transmitted or scattered sound at one frequency, while the angle is varied, and then repeating the process at each of a series of discrete frequencies, it is more convenient to sweep both variables simultaneously at appropriate rates. Typically, the frequency is swept uniformly through a range, e.g., from 0.8 to 1.0 MHz, every 10 sec, while the angle changes continuously by 1°. Since the acoustic signal is generated as short bursts, with a repetition rate of 10 sec^{-1}, the measurements for one quadrant of angular variation may be represented as a regular two-dimensional array of 9000 intensities, like a slightly skewed television picture. The angular distribution of the scattered sound may be treated similarly. The signal from the acoustic detector is heterodyned with a swept reference frequency in order to permit narrow-band amplification.

M10. Asymmetries in Cylindrical Waveguides. G. Z. FORRISTALL AND J. D. INGRAM, *Rice University, Houston, Texas 77001.*—Cylindrical geometrics with axial symmetry are often used to model physical systems. Although such a model materially simplifies calculations, it may ignore significant effects arising from small asymmetries in the system. In order to demonstrate this and to examine the nature of such effects, a model was analyzed consisting of a cylindrical waveguide with perfectly reflecting walls and an isolated point source displaced a small distance from the axis of the hole. The subsequent analysis shows two distinct types of arrivals associated with the geometry. Both of these produce strikingly large asymmetries in the motion of the system. These effects are clearly displayed in the closed-form solutions obtained for the problem, and the light that they shed on the nature of the reflection process at a cylindrical interface may well be of significant value for ray theory methods in diffraction.

M11. Model Studies on an Array of Interacting Piston Radiators. GEORGE J. GRUBER, *College of Engineering, University of Texas, Austin, Texas 78712.*—The mutual radiation impedance has been studied as a function of separation distance, element phasing, and shading using a model array of circular piston projectors radiating into a pool at 18 kHz. The elements are acoustically small ($ka=0.72$) and of the high Q-type ($Q=120$), mounted in an effectively infinite, rigid baffle. The sound generators consist of slender rods driven into their fundamental resonance mode by ceramic tubes, conveniently provided with vibration pickups for the measurements of the head velocities. The direct mechanoacoustical methods, in contrast to previous electroacoustical techniques, used to determine the absolute values of both the real and the reactive parts of the acoustical coupling coefficients, the induced pressure- and amplitude-distributions over the array, and the phase shifts needed at the electrical terminals of the vibrating pistons for maximum radiated power, are discussed. Reported *network* computations based on known equivalent mechanical circuit representations and iterative-type *field* solutions for interacting active–passive and active–active elements are both in agreement with the experimental data. [Work supported by the National Science Foundation, Engineering Division, and the U. S. Office of Naval Research, Acoustics Programs.]

M12. Initial Value Problems for Acoustic-Gravity Waves. ALLAN D. PIERCE, *Massachusetts Institute of Technology, Cambridge, Massachusetts 02139.*—Small-amplitude acoustic-gravity waves are governed by the linearized equations of hydrodynamics with gravity included and with the ambient atmosphere in hydrostatic equilibrium. The analog of Green's theorem for these equations is derived and applied to the formulation of the initial value problem solution. Although five different types of point sources must be considered in general, it is shown that one Green's function suffices if the ambient atmosphere is isothermal and at rest. The transient solution for waves from a spherical source initially motionless and with ambient density, but at higher than ambient pressure, is derived in terms of an integral over frequency and volume. The integral is evaluated near times of first arrivals and approximately evaluated at other cases of interest using the saddle point approximation. Numerical techniques are discussed and applied to the calculation of typical waveforms. At early times the results agree with the N-wave profile derived by Baker and Copson for a spherical wave in a homogeneous medium without gravity. The effect of gravity is to add a long period oscillatory tail to the waveforms.

M13. Radiation from a Spherical Array Enclosed by a Concentric Spherical Shell. R. E. DOUGLASS, B. E. JAY, AND W. C. MOYER, *TRACOR, Inc., Austin, Texas.*—Farfield beam patterns and mutual acoustic impedances have been computed for circular pistons mounted in a spherical baffle that is enclosed by an elastic spherical shell. For the beam pattern computation, the pistons are arranged in a rectangular array and phased to a plane to direct a narrow beam. The effect of enclosing the array by the shell is investigated by comparing with the no-shell results. It is seen that the shell does not change the main lobe of the beam pattern, but that the side lobe level is raised. The interaction coefficient between element pairs is given as a continuous function of element spacing, with zero spacing corresponding to the self-impedance. While addition of the dome does not change the basic shape of these curves, which are decaying oscillatory function, the shell-baffle spacing is seen to have a strong effect on the magnitudes of the interaction coefficients.

WEDNESDAY, 15 NOVEMBER — BURGUNDY EAST, 9:00 A.M.

Session N. Psychological and Physiological Acoustics III: Symposium on Psychophysiology of Binaural Hearing

M. H. GOLDSTEIN, *Chairman*

W. D. NEFF, *Discussant*

Invited Papers (25 minutes)

N1. The Morphology of Auditory Neurons and Auditory Localization. J. M. HARRISON, *Psychological Laboratory, Boston University, Boston, Massachusetts, and New England Regional Primate Center.*—The mammalian auditory system of the brain stem contains a number of classes of nerve cells the morphology of which may be relevant to the binaural localization of sounds. The axons of the acoustic nerve give rise to large synaptic endings (bulbs of Held) on the cell bodies of two types of nerve cells (types *c* and *g*) of the cochlear nucleus. The size of these endings suggests that they dominate the firing patterns of the postsynaptic cells, preserving in them the temporal characteristics of the activity in the acoustic nerve fibers. The medial superior olivary (mso) nucleus contains horizontally oriented bipolar nerve cells whose lateral dendrites probably receive synaptic endings from contralateral type *c* cells. Their bilateral innervation from cells that are synaptically dominated by the acoustic nerve fibers has suggested that the cells of the mso are of significance in the location of sounds. The medial nucleus of the trapezoid body contains nerve cells each of which is innervated by a single massive synaptic ending (chalice of Held) from large axons of the contralateral type *g* cells. This would appear to be a rapidly conducting system in which temporal characteristics of acoustic nerve activity may be preserved. The number of nerve cells in the major nuclei of the superior olivary complex varies across mammalian species. The mso varies from 0 (bats and dolphin) to 4200 (cat) nerve cells. The absence of this nucleus in the echolocating bats and dolphin shows that this is not an essential structure for sound localization in mammals. The lateral superior olivary nucleus is well developed in bats (4300) and dolphin and is also present in all species we have examined. This suggests that this nucleus is involved in auditory localization and complex auditory discriminations. The medial nucleus of the trapezoid body is well developed (2100–5800 nerve cells) in all species examined.

N2. Electrophysiological Studies of Binaural Hearing. JAY M. GOLDBERG, *Department of Physiology, University of Chicago, Chicago, Illinois.*—Attention is focused on the superior olivary complex. Of the nuclei of the complex, the medial superior olive (MSO) is unique in that most of its neurons are affected by stimulation of either ear. Three-quarters of the binaural MSO neurons—*EE cells*—receive a predominantly excitatory input from both ears. The remainder—*EI cells*—are activated by stimulation of one ear and inhibited by stimulation of the other ear. The response of MSO neurons to dichotic stimuli is reviewed and compared with the behavior of cells in other nuclei. *Response to clicks: EI* cells are sensitive to interaural time and intensity differences and show time-intensity trading. *EE* cells respond to binaural clicks, usually in an occlusive manner. *Response to tones: EE* cells are sensitive to the average intensity of dichotically presented tones and are relatively insensitive to interaural intensity differences. *EI* cells may be insensitive to average intensity, but quite sensitive to interaural intensity differences. Both *EE* and *EI* cells respond to interaural phase differences; such cells discharge in a phase-locked manner to low-frequency sinusoids. [Work supported by National Institutes of Health grant.]

N3. Binaural Hearing in Cat and Man. IRVING T. DIAMOND, *Department of Psychology, Duke University, Durham, North Carolina.*—Cats were trained to respond in a double-grill box to a change in a binaural pattern of tones produced by alternately stimulating one and then the opposite ear. Tests showed that the cats did not, in fact, perceive a single sequence to which each ear contributed its share, but instead waited for a frequency change to occur in one ear. Ablation of auditory cortex contralateral to the side of the attentive ear resulted in an amnesia, while ablation of the auditory cortex ipsilateral to the attentive side had no effect upon the learned habit. The question of how the engram is isolated to one hemisphere was explored by further studies in which the corpus callosum was sectioned. The results suggest that the engram is kept out of the ipsilateral hemisphere by an active suppression originating in the contralateral hemisphere. In the course of these experiments, it was observed that the rate of clicks that alternate from side to side was underestimated when compared to a train of clicks presented monaurally. A subsequent experiment by Axelrod with college students as subjects showed precisely the degree of underestimation. [Work supported by Grant from N. I. M. H.]

N4. Binaural Effects in Psychoacoustic Research. DAVID M. GREEN, *University of California at San Diego, La Jolla, California 92038.*—A review of the psychoacoustic research on sound localization and the role of phase and amplitude differences between the two ears is presented. The precedence effect and the steady-state cues for localization are related. A survey of the work on masking level differences (MLD's) as well as two prominent theoretical accounts of this phenomena are discussed. [Research supported in part by the National Institutes of Health, Public Health Service, U. S. Department of Health, Education and Welfare.]

N5. On Listening to Competing Stimuli and Functional Differentiation of the Cerebral Hemispheres. DONALD SHANKWEILER, *Haskins Laboratories, New York, New York.*—A number of experiments have indicated a reliable right-ear advantage in recognition of speech materials presented dichotically to the two ears. If the stimuli are melodic patterns, however, listeners recognize more accurately those presented to the left ear. Since each ear has greater representation in the opposite cerebral hemisphere, these lateral differences in perception must reflect functional differences between the hemispheres in processing speech and nonspeech stimuli. The special relation of the left cerebral hemisphere to language processes has long been recognized, but the details of this relationship have remained obscure, partly because the evidence could only be obtained through study of patients who had incurred brain damage. The technique of dichotic listening offers an approach to the further specification of the functions of the two cerebral hemispheres in persons with intact nervous systems. Experiments with speech materials are described that bear on the problem of the units of perception and the procedures by which the perceptual apparatus arrives at phonemic decisions.

WEDNESDAY, 15 NOVEMBER — MEDALLION EAST, 9:00 A.M.

Session O. Underwater Acoustics III: Symposium on Adaptive Signal Processing

F. V. HUNT, *Chairman*

Invited Papers (30 minutes)

O1. Estimation and Detection: A Unified Approach I. HENRY COX, *Naval Ship System Command, Washington, D. C. 20360.*—The thesis of this paper is that a variety of results in optimum detection and estimation can be most easily comprehended if one first examines in detail some very simple but general problems and fully understands the assumptions, solutions, and interrelationships for these simple problems. The relationships among a variety of seemingly diverse, complicated problems may then be clearly seen since the first step in obtaining a solution is usually to transform the problem by some technique into one of the simple problems. The transformation step is frequently of such mathematical complexity that the basic strategy is effectively camouflaged. After examining the simple general problems, various applications, including optimum arrays, are discussed.

O2. Null Steering as a Realizable Process and Its Relationship to Optimum Beamforming. VICTOR C. ANDERSON, *University of California, San Diego, Marine Physical Laboratory of the Scripps Institution of Oceanography, San Diego, California 92152.*—A real-time digital processor is described that is designed to reject plane-wave interference from an operator-controlled direction, prior to normal beamforming. Both hardware and an analysis are discussed. The process is compared with the corresponding optimum process, indicating the similarities and differences that occur.

O3. Adaptive Antenna Systems. B. WIDROW, P. MANTEY, L. GRIFFITHS, B. GOODE, *Electrical Engineering Department, Stanford University, Stanford, California.*—A system consisting of an antenna array and an adaptive processor can perform filtering in both the space and frequency domains, thus reducing the sensitivity of the signal-receiving system to interfering directional noise sources. Variable weights of a signal processor can be automatically adjusted by a simple adaptive technique based on the least-mean-squares (LMS) algorithm. During the adaptive process an injected pilot signal simulates a received signal from a desired "look" direction. This allows the array to be "trained" so that its directivity pattern has a main lobe in the previously specified look direction. At the same time, the array processing system can reject any incident noises, whose directions of propagation are different from the desired look direction, by forming appropriate nulls in the antenna directivity pattern. The array adapts itself to form a main lobe, with its direction and bandwidth determined by the pilot signal, and to reject signals or noises occurring outside the main lobe as well as possible in the minimum mean-square error sense. Several examples illustrate the convergence of the LMS adaptation procedure to the corresponding Wiener-optimum solutions. Rates of adaptation and misadjustments of the solutions are predicted theoretically and checked experimentally. Substantial reductions in noise reception are demonstrated in computer-simulated experiments. The techniques described in this paper are applicable to signal-receiving arrays for use over a wide range of frequencies.

O4. Adaptive Nonparametric Detection. HERBERT L. GROGINSKY, *Raytheon Company, Wayland, Massachusetts.*—The aim of an adaptive detector is to vary the characteristics of the device so that nearly optimal sensitivity is obtained among classes of signals from targets or noises perturbing the measurements. The objective is to reduce the losses inherent in using Bayes average statistics to describe the signals and noise. In order to obtain this functional independence, two sensors are used, only one of which can contain the signal. The technique then is to choose a processing system that is inherently distribution free (nonparametric) and vary the available free parameters of the processor in accordance with the system inputs, to bring the sensitivity of the detector to nearly that which could be obtained if the statistics of both the signal and noise were known. It can be shown that if an optimal nonparametric detector exists it must be of the rank-order type. For weak signals, the optimum detector uses linear processing of the rankings. In the adaptive detector described in this paper, the linear processor coefficients are varied intermittently in accordance with the results of each successive detection test. Theoretical calculations verified by computer simulation indicate that this type of device has a definitive lower threshold so that no detection can occur at all with signals below the threshold, while for signals above threshold, essentially perfect detection can take place given sufficient data.

O5. The Adaptive Likelihood Ratio Detector. THEODORE G. BIRDSALL, *Cooley Electronics Laboratory, The University of Michigan, Ann Arbor, Michigan, 48105.*—A search for a theory of optimum adaptive detectors leads instead to a theory of adaptive realizations for optimum (likelihood ratio) detectors. The virtue of adaptation is not in achieving breakthroughs in performance. The real role of adaptation is that certain levels of performance that are achievable by both adaptive and nonadaptive detectors may be achievable with cheaper or smaller equipments if they are adaptive. The paper ends with a short discussion of implementation of the design rules furnished by likelihood-ratio theory.

WEDNESDAY, 15 NOVEMBER — MEDALLION WEST, 2:00 P.M.

Session P. Noise III: Community Noise, State of the Art

R. R. AUDETTE, *Chairman*

Invited Papers (25 minutes)

P1. Airborne Transportation Noise—Its Origin and Abatement. JOHN O. POWERS, *Office of Noise Abatement, Federal Aviation Administration, Washington, D. C.*—The growth of airborne transportation has resulted in an increased awareness of the sounds associated with this form of travel. The noise generated by propeller-driven aircraft is given way to the sound of the exhaust jet, the compressor, and the fan whine. In addition, with the advent of supersonic flight, a new sound, the sonic boom, has been introduced to our acoustic environment. It is anticipated that airborne transportation will increase markedly in the coming years and as a result there could be a marked increase in the noise problem. The Federal Aviation Administration is attempting to minimize the acoustic impact of this increased air travel by attacking the problem on many fronts. The problem is initially defined through the mechanism of noise exposure forecasting. Methods are continually sought to reduce the noise at the source, to alter the sound transmission path, and to protect the community from the acoustic intrusion. Finally, procedures for national and international certification and regulation are being explored as a means of noise control.

P2. Surface Transportation. III. G. J. THIESSEN, *National Research Council, Ottawa, Ontario.*—As technology advances, the level of noise in our everyday environment appears to increase. Among the factors contributing to this in the traffic field are the increased horsepower used in vehicles, the reduction in weight of engines, the higher compression ratios, and the increased use (especially in buses and trucks) of diesel engines. Legislative control has so far been somewhat timid and lacking the followthrough of enforcement. Noise-control technology has been applied only sporadically, so that the range of noise commonly encountered in any octave band for standard-size passenger cars is more than 10 dB. For heavy trucks, it is even greater. Motorcycles at full throttle, almost regardless of speed, exceed the noise of tractor trailors crusing at 60 mph. Stationary baffles have been successfully used, especially for trains. Rubber tires have also been found practical in subways and urban trains. Proper highway location, design, and noise zoning are further means toward a quieter community.

P3. Industry Noise. KENNETH M. MORSE, *Director of Industrial Hygiene, United States Steel Corporation, Pittsburgh, Pennsylvania.*—The definition of noise as "unwanted sound" is purposely devoid of technical language owing to the wide spectrum of possible effects upon humans that are both physiological and psychological. The former is largely objective and the latter subjective. While there are a considerable number of published efforts to establish criteria on hearing-damage risk, such "numbers" are widely debated upon the basis of lack of valid supporting data. In the area of annoyance, efforts to establish criteria are on even more untenable grounds. Attempts to equate degrees of annoyance are most difficult and highly complex. An effort to attach numbers to annoyance is faced not only with emotional and sociological factors but also with the fact that there are wide differences in human susceptibility and adaptability to noise. This paper discusses the major efforts that have been made to evaluate annoyance from industrial noise, zoning laws with performance standards, the rôle of transportation noise in such standards, and the views of the author on the usefulness of such data.

P4. Air Conditioner Equipment. A. E. MELING, *ARI, Arlington, Virginia.*—Although early residential air conditioners were usually water-cooled, air-cooled versions, which may result in complaints if not properly controlled, quickly became by far the dominant type. The first efforts to control such noise by means of ordinances took place about 1958. These efforts continued on a quite sporadic basis until the summer of 1966 when they mushroomed in such widely separated suburbs as Coral Gables (Miami), Fla., Beverly Hills (Los Angeles), Calif., Irvington (New York), N.Y., and even Etobicoke (Toronto), Ontario. Since sound tends to be confusing to people who are not experts in acoustics, it is not surprising that these noise ordinances vary from effective to unenforceable. To fill an obvious need in this regard, ARI made a study of the factors that should be taken into account by communities considering such ordinances. The resulting "model ordinance" is designed to control noise that will be a nuisance to most people and is practical to enforce in the field. The ARI ordinance is not a panacea for this problem. The real solution will require a coordinated blend of good equipment, proper installation practices, and realistic controls.

P5. The Noise Receiver, the Citizen. ROBERT ALEX BARON, *Citizens for a Quieter City, Inc., New York, New York 10036.*—Urban man, immersed in everyday noise, receives little benefit from the science and technology of acoustics. He is low man in the noise source-transmission path-receiver system. Yet there is extensive know-how for noise control in industry and aerospace. In contrast, urban man continues to suffer, unabated and magnified, the same disabling noises of two generations ago. Today in addition, his environment is blighted by the inadequately muffled noises of aviation. Uniform guidelines are needed. Faced with this indisputable evidence of neglect, urban man must become responsible for his own protection. Citizens for a Quieter City has been formed to create an awareness of the need for the control of city noises and of the means available for noise control. It has a formal structure and draws upon a wide spectrum of experience and support, including that of humanist scientists and technologists. It is concluded that a knowledgeable citizen's noise abatement organization is a necessity. CQC has already proven productive, and extensive planning for the future is underway. A quieter everyday environment is possible, reasonable—and long overdue.

P6. Criteria for Control of Community Noise: a Candid Appraisal of Some Unresolved Questions as seen by One Acoustical Consultant. WARREN E. BLAZIER, JR., *Bolt Beranek and Newman Inc., San Francisco, California.*—The acoustical consultant frequently finds himself on both sides of the problem in situations concerned with community noise. In either case, he must develop a criterion to evaluate the particular problem area—considering not only the physical control measures that are practical relative to the type source, but also the probable noise reduction requirements to achieve the subjective approval of the receiver. Each situation is likely to be complex regardless of whether the objective is to *prevent* a complaint or to correct one that *already* exists. There is a tendency on the part of municipal governments and other regulating agencies to establish noise limits on a basis which is too generalized or oversimplified to handle the broad range of individual situations encountered. An example is the ordinance of Coral Gables, Florida, which limits the noise level of residential air conditioning equipment installed outdoors. A comparison is made between this ordinance and noise data obtained on a large number of units operating, without complaint, in that city during the summer of 1966. General comments are made in connection with this survey which bear on the more fundamental problems faced in developing objective criteria for control of community noise.

WEDNESDAY, 15 NOVEMBER — MEDALLION EAST, 2:00 P.M.

Session Q. Physical Acoustics III

M. A. BREAZEALE, *Chairman*

Invited Papers (25 minutes)

Q1. Acoustic Excitation of Nuclear Spin Resonance in Solids. E. H. GREGORY, *Department of Physics, California Institute of Technology, Pasadena, California.*

Q2. Limitations on the Performance of Ultrasonic Delay Lines Using Vitreous Silica as the Delay Medium. A. H. MEITZLER, *Bell Telephone Laboratories, Murray Hill, New Jersey.*—The information storage capabilities of ultrasonic delay lines using vitreous silica (a term used to denote both fused quartz made from powdered crystalline quartz and fused silica made by oxidation of $SiCl_4$ in an oxygen–hydrogen flame) are limited at low frequencies by the variation in the shear wave phase velocity and at high frequencies by the attenuation-vs-frequency dependence of vitreous silica. Recent experimental data [D. B. Fraser, J. T. Krause, and A. H. Meitzler, Applied Physics Letters (In preparation)] on the shear attenuation coefficient versus frequency dependence and the constancy of shear wave phase velocity are presented and combined with analytical models to estimate the maximum digital signal storage capabilities of ultrasonic delay lines using vitreous silica. In addition, other factors influencing performance limits, such as the temperature dependence of loss, the temperature coefficient of delay time, and the magnitude of the acoustoelastic effect in vitreous silica, are discussed.

Q3. Twist Overtones of Thickness-Shear and Flexure in *AT*-Cut Quartz Plates with Partial Electrodes. P. C. Y. LEE, *Department of Civil Engineering, Princeton University,* AND W. J. SPENCER, *Bell Telephone Laboratories, Inc., Allentown, Pennsylvania.*—The equations developed by Mindlin and Spencer for twist overtones of thickness-shear and flexure have been applied to *AT*-cut quartz plates with partial electrodes. The frequency and displacements for resonances near the fundamental thickness-shear mode have been calculated. These results are compared with measured resonances and the mode shapes determined by x-ray diffraction topography.

Q4. Ultrasonic Coupling Loss. WALTER L. GHERING, *Pennsylvania State University, University Park, Pennsylvania.*—Ultrasonic coupling losses are found to be a significant component of the experimentally observed loss in measurements of small, high Q specimens. An investigation of the characteristics of this coupling loss showed that it is mainly a dynamic viscous-type loss or a surface friction loss. Viscous losses due to shear components of the vibration are found to be important for "longitudinally" vibrating disks. An analysis of the Aggarwal solution for the disk shows the shear component of the energy to be of the same order of magnitude as the longitudinal component. Experimental data show the loss due to free films, films loaded with $\frac{1}{2}$ and $\frac{1}{4}$ wavelength samples, and the coupling loss in optical contact. A variation of the technique used to investigate coupling losses appears to have promise for dynamic viscosity measurements as a function of frequency. [This work was supported by Air Force Grant.]

Q5. Acoustic Emission from Metal Caused by Changes in Applied Stress. LARRY MITCHELL AND J. N. KERAWALLA (nonmember), *E. I. du Pont de Nemours, Wilmington, Delaware* AND J. R. FREDERICK, *University of Michigan, Ann Arbor, Michigan.*—When the stress level on a metal specimen is increased or decreased, it is possible to observe bursts of noise by the use of a sensitive contact microphone. At stress levels within the elastic range, the emission observed during the removal of the stress is usually at least two orders of magnitude greater than when it is being applied. When the applied load is removed, there is no emission until the stress has decreased by an amount Δs from the maximum load. The magnitude of the emission is dependent upon the strain rate and on the metallurgical and stress history of the specimen. [Work supported in part by the National Science Foundation.]

Q6. Bechmann's Number for an Inverted Mesa Structure. J. L. BLEUSTEIN, *Yale University, New Haven, Connecticut.*—If the ratio of the length of the electrode strip to the thickness of a partially electroded infinite crystal plate is kept below a certain critical value, Bechmann's number, then, for the crystal vibrating at frequencies in the neighborhood of its fundamental thickness-shear frequency, the current through the crystal will be unaffected by the anharmonic overtones of the thickness-shear motion. The concept of the inverted mesa structure (a crystal with an electrode and an insulating film) can be employed to design crystals with large values of Bechmann's number without increasing the electrical resistance of the electrode. In this paper, by means of a special approximate theory, an explicit formula for the Bechmann's number of an inverted mesa structure, in terms of the dimensions and the material properties of the insulating film the electrode and the crystal, is obtained.

Q7. Sound Generation by a Dusty Gas. D. G. CRIGHTON, *Department of Mathematics, Imperial College, London, England.*—When a finite region of fluid is in turbulent motion, the stresses in the fluid generate sound waves in the surrounding fluid in the same manner as a suitable distribution of acoustic quadrupoles occupying the trubulent region. If the turbulent fluid also contains many small dust particles, the viscous drag force between the dust and gas will also cause generation of sound, but in the manner of a distribution of dipoles. At sufficiently low Mach numbers, the dipole-generated sound must dominate the radiation from the turbulence, and it is shown that this will be true for Mach numbers less than f, the mean dust mass concentration ratio. [Research carried out under the Bureau of Ships and General Hydromechanics Research Program, administered by the David Taylor Model Basin, under contract.]

Q8. The Effect of the Acoustic Fresnel Region of Transducers on Optical Measurements. F. INGENITO, A. J. CRANDALL, AND BILL D. COOK, *Department of Physics, Michigan State University, East Lansing, Michigan.*—The optical techniques for the investigation of ultrasonic waves have been limited by the lack of an exact mathematical expression relating the velocity amplitude at the source to the optical phase retardation experienced by the light passing through the sound beam. For low frequencies, (i. e., Raman Nath region) general expressions for the optical phase retardation for plane radiators of arbitrary geometry in a rigid plane baffle have been developed. Results for circular and rectangular sources are presented. [Work supported by the Office of Naval Research.]

Q9. Variable Angle Spectrometer for Brillouin Scattering (SBS). C. LEONARD O'CONNOR, *Department of Physics, Manhattan College, Bronx, New York 10471.*—An experimental system has been designed to study spontaneous Brillouin scattering from liquids with scattering angles that can vary continuously from 20° to 160°. This range of angles affords a method for velocity and absorption measurements to be made over a substantial portion of the dispersion region in most liquids, namely, from about 0.6 to 6.0 Gc/sec. The corresponding analyzing and detecting systems for the scattered radiation have been described earlier [J. Acoust. Soc. Am. 40, 663 (1966)]. The scattering cell is cylindrical and temperature controlled to 0.1C°. A variable cross arm, glass recticle, and autocollimation system in conjunction with an optical spectrometer is used to align and measure the scattering angle to at least $\pm 0.1°$. Experimental results using a selection of organic liquids will be compared with data previously obtained by ultrasonic techniques.

Q10. Light Scattering from Thermally Excited Surface Waves. ROBERT H. KATYL (nonmember) AND UNO INGARD, *Department of Physics and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, Massachusetts.*—We report results of a spectral analysis of light scattered by a liquid surface. These results are consistent with the assumption that thermally excited surface waves are responsible for the scattering, and the phenomenon may be regarded as the surface-wave analogue of ordinary Brillouin scattering in the bulk of a fluid. [This work was supported principally by the U. S. Navy Office of Naval Research under Contract and in part by the Joint Services Electronics Program; additional support was received (by R. H. K.) through a National Science Foundation Graduate Fellowship.]

Q11. Out-of-Plane Motion of a Stretched String under Forced Vibration. D. W. OPLINGER, *Avco Corporation, Space Systems Division, Lowell, Massachusetts.*—This paper treats the nonlinear phenomenon of out-of-plane motion that may develop under certain conditions in a stretched string subjected to planar forced vibrations. As in a previous paper dealing with the case of in-plane motion [J. Acoust. Soc. Am. (Dec. 1960)], the lack of inertial effects for longitudinal deformations occuring during transverse vibrations leads to tension levels that are uniform along the string but varying in time, and separable solutions are seen to result. In the course of the discussion, the out-of-plane motion is analyzed in terms of stability diagrams of Mathieu's equation, and results provide an interpretation of two types of out-plane motion, a "jump rope" mode (out of plane component 90° out of phase with in-plane component) and a "warped plane" mode (out of plane component in phase) which have been observed in experiments involving a slotted end constraint on the string. In the oral presentation, a motion picture sequence illustrating the characteristics of transverse vibration in an actual vibrating string will be presented.

Q12. Ultrasonic Method for Measuring the Thermal Expansion of Liquids. J. H. PATTISON, *David Taylor Model Basin, Washington, D. C.*, AND J. JARZYNSKI AND C. M. DAVIS, *Physics Department, The American University, Washington, D. C.*—A new method of measuring the coefficient of thermal expansion of a liquid is described. First, the velocity of sound is measured over a precisely known fixed path length. Next, the time of flight of an ultrasonic pulse reflected from the free surface of the liquid is determined as a function of temperature. Using this data, the coefficient of thermal expansion is calculated. Preliminary measurements in mercury show that the accuracy of the accuracy of the above technique is $\pm 1\%$. Furthermore, the combination of ultrasonic velocity and thermal expansion data permits calculation of the isothermal compressibility. [Work supported by a grant from the National Science Foundation.]

WEDNESDAY, 15 NOVEMBER 1967 — SILVER CHIMES EAST, 2:00 P.M.

Session R. Psychological and Physiological Acoustics IV: Psychoacoustics of Binaural Hearing

CHARLES WATSON, *Chairman*

Contributed Papers (10 minutes)

R1. The Haas and/or Precedence Effects in Sound Localization. MARK B. GARDNER, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—When two or more sound sources radiate identical complex signals, the listener tends to hear only a single image located at the nearer source, providing the maximum delay is less than about 50 msec. This phenomenon (together with other closely associated aspects) is generally referred to as the "precedence effect" or Haas effect." Less frequently, the terms "Law of the First Wave Front," "auditory suppression effect," and "threshold of extinction" have also been used in describing certain of these characteristics. Perhaps the earliest investigator to leave a clear record of his appreciation of the single image aspect of the phenomenon was Joseph Henry who, in 1849, used the term "limit of perceptibility" in describing what he had observed. The present paper will give some of the historical background of the effect primarily from the standpoint of its investigation and its application in electro-acoustical systems. In addition, some recent data on the influence of multiple equal-energy sources on the accuracy of localizing the apparent single source will be given.

R2. Free-Field Localization of Acoustic-Images. F. E. TOOLE (nonmember), *Division of Applied Physics, National Research Council, Ottawa 7, Ontario, Canada.*—Experiments in free-field localization where head movements are restricted have revealed that complex in-head and external localizations frequently occur. The in-head localization may be total (including all of the sound) or partial, in which case part of the sound is localized somewhere in the space around the listener. The externalized sound may be localized correctly in the direction of the loudspeaker, or with the same degree of sidedness but with front-back reversal, or at both correct and reversed locations simultaneously. The factors giving rise to localization judgments in each of these categories are not immediately obvious, and the situation is complicated by intersubject differences. It is possible, however, to observe systematic changes in localization as a function of certain signal parameters such as frequency and bandwidth. These trends may provide clues to the nature of the auditory mechanism underlying front-back discrimination.

R3. Time-Intensity Trade and the Lateralized Location. CARL W. ASP, *University of Tennessee, Knoxville, Tennessee, 37916.*—Twenty-six male listeners completed the time–intensity trade and judged the lateralized location of interaural intensity differences from reference conditions of binaural loudness balance and binaural intensity balance. These signals of 150, 300, 600, and 1200 Hz were interrupted and had burst durations of 27, 20, 13, and 11 msec, respectively. The mean trading ratio increased as the frequency decreased and as the interaural intensity-difference increased. At lower frequencies, the mean ratio was larger when the trade was completed in a right-lateralized position. The mean judgments of lateralized location were 3.5 and 9.4 for right and left attenuation of 15 dB, respectively. The latter mean had a greater absolute difference from the midline number of 6.0. This indicated more displacement toward the right ear.

R4. Detection and Localization: Analysis of an Ideal Observer. RAMON L. HERSHMAN AND M. LICHTENSTEIN, *Naval Electronics Laboratory Center, San Diego, California 92152.*—We extend the theory of signal detectability to a problem in detection and localization (DAL). On each trial, at most one of k observation intervals contains a signal, and the observer indicates either "no signal present" or "the signal is in interval

j" ($1 \leq j \leq k$). To maximize correct decisions, the ideal DAL observer is realized by the sequential application (in either order) of YES–NO aid Forced-Choice (FC) decision rules. The theory predicts a higher percent correct on signal trials than on NO-signal trials. Quite independently, the observer has an apparent decision "bias" for saying "signal." Ideal DAL and FC receivers are compared.

R5. Lateralization of an Auditory Signal in Correlated Noise and in Uncorrelated Noise as a Function of Signal Frequency. DONALD E. ROBINSON AND JAMES P. EGAN, *Hearing and Communication Laboratory and Department of Psychology, Indiana University, Bloomington, Indiana.*—Listeners lateralized a monaural signal presented against a continuous background of perfectly correlated noise (NO) and of uncorrelated noise (NU). Measures of signal detection were also secured in separate tests. Psychometric functions (percent correct versus signal energy) were determined for each task. For a tonal signal of either low or high frequency, the listener requires only slightly greater signal energy (1–2 dB) in order to lateralize as well as he can detect when the noise is uncorrelated (NU). When the noise is perfectly correlated (NO), the slope of the psychometric function for lateralization depends upon signal frequency. With 250 cps, the slope of the function for lateralization is much smaller than that for detection. With 1000 cps, the function for lateralization is steeper than that for 250, but the slope is still less than that for the function for detection of 1000. With 2000 cps, the function has about the same slope as that for detection. [Research supported by U. S. Air Force Office of Scientific Research.]

R6. Negative MLD's with Gated Sinewave Masking. FREDERIC L. WIGHTMAN, *University of California, San Diego, La Jolla, California 92037.*—The detectability of S_0 and S_π signals, masked by a tone of the same frequency and duration, was measured as a function of the signal-masker phase. The signal and masker frequency was 250 Hz, and the duration 1/10 sec. In all conditions, S_π signals were less detectable than S_0 signals (a negative MLD), and this effect was largest (about 10 dB) with signal and masker in phase. Detection of S_0 signals as a function of signal-masker phase closely follows the predictions of a simple energy detection model. The results with S_π signals, however, are not consistent with any existing theory of binaural detection. In some general respects, the data suggest a mechanism similar to Durlack's equalization–cancellation mechanism. [Research supported in part by the National Institutes of Health, Public Health Service, U. S. Department of Health, Education and Welfare.]

R7. Masking-Level Differences Determined with and without Interaural Disparities in Masker Intensity. DENNIS MCFADDEN, *Department of Psychology and Defense Research Laboratory, The University of Texas at Austin, Austin, Texas 78712.*—Psychometric functions were obtained for three listeners for the interaural conditions Nm–Sm and NO–S_π using seven different noise spectrum levels. In addition, data were collected on the NO–S_π condition with the spectrum level in one ear smaller than that in the other ear. In these NO′–S_π′ conditions, the signal-to-noise ratio was kept equal in the two ears. The masking-level differences (MLD,s) for the NO–S_π conditions declined gradually as the level of the noise was reduced, and for small spectrum levels were essentially the same as the MLD obtained when the masker was turned off. The MLD,s declined more rapidly for the NO′–S_π′ conditions, but a small MLD was obtained even when the spectrum level in one ear was 60 dB smaller than that in the other ear. For the Nm–Sm conditions, masking increased linearly with increases in spectrum level, but for NO–S_π the relation was not linear. Signal: 400 cps, 250 msec. Masker: continuous, wide-band noise. Method: 2IFC. [Research supported in part by Air Force Office of Scientific Research and Public Health Service.]

R8. Dichotic Loudness Summation for Pure Tones. BERTRAM SCHARF, *Department of Psychology, Northeastern University, Boston, Massachusetts 02115.*—The over-all loudness of two equally loud tones, one presented to the left ear, the other to the right ear, was measured as a function of their frequency separation. The two tones were presented simultaneously for 700 msec and matched in loudness to a 2000-Hz binaural tone. Matches were made at 20, 50, and 80 dB above the threshold for the binaural tone. The geometric mean of the dichotic tones was 2000 Hz; seven frequency separations were used, from 100 Hz (one-third of a critical band) to 4530 Hz (12 critical bands). Results from six subjects indicated that the loudness of the two tones presented dichotically is independent of frequency separation. Moreover, at all three sensation levels dichotic loudness summation is the same quantitatively as binaural loudness summation. On the average, a dichotic or binaural sound is about as loud as a monaural tone that is 8 dB more intense. [Work supported by a grant from the U. S. Public Health Service, National Institute of Neurological Diseases and Blindness.]

R9. On Central Masking and a Time Constant of the Auditory System. J. J. ZWISLOCKI AND R. A. NEWMAN, *Laboratory of Sensory Communication, Syracuse University, Syracuse, New York.*—Previous experiments performed at 1 kHz have been extended to 2 and 0.25 kHz. The new results together with the preceding ones lead to the following conclusions. The effect of central masking (as measured in terms of threshold shift) is maximum at the time of masker onset. The maximum increases with sound intensity, but remains frequency independent when the loudness level in the critical band under investigation is kept constant. Provided the latter is true, the maximum does not depend on spectral characteristics, i. e., whether the masker is a pure tone, narrow-band, or wide-band, or wide-band noise. The threshold shift decays exponentially as the time delay from the masker onset is increased; it reaches an asymptote at about 250 msec. The time constant of the decay, on the order of 50 msec, appears to be independent of stimulus parameters. It may be considered a a system,s constant. The same time constant may be derived from various neurophysiological recordings.

R10. Short-Term Loudness Changes. ARNOLD M. SMALL, JR., *University of Iowa, Iowa City, Iowa.*—In a previous study, it was reported that in a forward masking situation loudness changes as measured by a median plane lateralization of the apparent sound source were vanishingly small [J. Acoust. Soc. Am. **38**, 928 (1965)]. This observation was somewhat surprising in view of the large threshold shifts seen using the same stimulus conditions. The present paper reports attempts to resolve this apparent paradox. The task of the six listeners was twofold: (1) to adjust the level at one ear of a binaurally presented test signal so that its sound image was congruent with an identical standard (except for the standard's fixed SPL). An adapting stimulus was interposed between the standard and the test signals. (2) The livel of both test signals was simultaneously adjusted so that their combined loudness was equal to that of the standard. The lateralization results show shifts of more than 20 dB with recovery proceeding nearly exponentially and being complete in 100 msec. The smallest shifts were observed at 1000 Hz with larger amounts at 250 and 4000 Hz. The direct loudness measures showed similar results, although they were smaller in magnitude. [This research was supported by National Science Foundation.]

R11. Binaural Unmasking and Frequency Discrimination. RONALD M. ROBERTSON AND DAVID P. GOLDSTEIN, *Purdue University, Lafayette, Indiana.*—Frequency difference limens (DL's) were determined for three observers employing a 300 Hz target signal for NOSO, NOSπ, and NmSm at three sensations levels (SL's) above masked threshold (5, 10, and 15 dB) and two masker levels (20- and 59-dB spectrum level). The size of the frequency difference limen in noise is dependent upon the sensation level at which it is measured. the interaural phase of the target signal and the level of the noise. The difference limen is significantly larger when measured at low sensation levels under high-level noise when the target signal is 180° out of phase. This effect diminishes at higher sensation levels or when noise level is reduced, and does not appear when the signal is in phase or when a monaural noise-signal configuration is used. Masker level has no effect on the size of the DL for NOSO or NmSm but a differential effect was seen for NOSπ at 5 and 10 dB SL under the higher masker level where larger DL's were observed as compared to the lower masker level. Theses results are predictable from previous findings in regard to the band width employed by the auditory system under conditions where masking level differences (MLD's) are observed.

WEDNESDAY, 15 NOVEMBER 1967 ROYAL ROOM, 2:00 P.M.

Session S. Speech II: Perception and Varia

HENRY M. TRUBY, *Chairman*

Contributed Papers

S1. Intonation: Accent and Emphasis, Cadence, and End-Glide. RALPH VANDERSLICE, *University of California, Los Angeles and RAND Corporation, Santa Monica, California.*—Lieberman's (1967) simplistic model of American intonation is wrong on three crucial counts: (1) it attributes most pitch change to variations in subglottal pressure, whereas EMG and cricothyrometry show that laryngeal adjustment is the primary physiological correlate; (2) it recognizes only one degree of prominence, whereas systematic synthesis requires *accent* [essentially Trager and Smith's primary and (prenuclear) secondary "stresses"] and *emphasis* (usually "overhigh" but with variants including Bolinger's "ac ent *C*"); (3) it conflates postnuclear pitch *cadence* with terminal pitch *endglide* although the domain of the former under context-dependent accent shift may be a whole order of magnitude longer than the postulated 150–200 msec. Distinguishing these sometimes overlapping gestures permits orderly treatment of traditional tunes in terms of binary features: unmarked (upward-obtruded) nuclear accent or emphasis may be followed by falling or nonfalling cadence (±Fall) with rising or nonrising endglide (±Rise). Intonations then are falling [+F, −R], rising [−F, +R], falling–rising [+F, +R], and sustained [−F, −R]. Major and minor types are distinguished by a third feature, ±Pause. [Work supported by Office of Naval Research.]

S2. Perception of Linguistic Boundaries. GEORGE W. HUGHES, ARTHUR S. HOUSE, AND JOHN A. RUPF, *Purdue University, Lafayette, Indiana.*—The necessity for segmentation in linguistic descriptions of an utterance has long been recognized. We have made a study of the perception of boundaries between phonemic segments of natural speech. Subjects were instructed to divide an utterance at the boundary between a consonant and vowel by causing the presentation (under phones) to switch from one ear to the other at a point in time corresponding to the CV or VC boundary. The means and variances of these subjective placements were measured and, intersubject variation was found to be greater than intrasubject variation, but both were large. Subjective boundaries seldom corresponded to prominent waveform features visible on a CRO. Subjective boundaries determined by switching were compared to subjective boundaries determined by listening to gated segments. [This work was supported in part by the Air Force Cambridge Research Laboratories under Contract.]

S3. Psychoacoustic Implications of Inter-Species Communication. HENRY M. TRUBY AND JOHN C. LILLY, *Communication Research Institute, Coconut Grove, Florida.*—As a corollary and prerequisite to the acoustic study of interspecies communication, consideration needs to be given to the psychoacoustic or psycholinguistic aspects of the problem. It should not, for example, be assumed that the communication systems (codes, "languages") of alien species bear resemblance to those of *Homo sapiens*, and they are very likely unlike any of the currently active 5000 mutually exclusive human languages. It is difficult enough for a given human speaker with a single native language to comprehend any language much unlike his own, in spite of the sharing, on both the transmitting and the receiving side, of similar frequency and time domains, as is true for all human languages, and he needs more than a little convincing that a plethora of languages contemporary with his own are by nature so radically different in structural design and coding that he could apply none of his inherent linguistic instincts to their comprehension. This leaves no opportunity for fundamental pattern or system comparisons, i.e., perhaps there is no lexical inventorying, no verbal system, no connectives, or the like; perhaps the temporal sequence is complexly coded; perhaps that which is basic to human linguistics is incidental or nonexistent in the codal organization of alien species. Consider, for example, the bottlenose dolphin (*Tursiops truncatus*), who has three sonic–ultrasonic emitters, two of which can be linked in double or stereo phonation, and the third of which is used for sonar operation. This species apparently has alternating cerebral dominance, high rates of body locomotion and other muscular-operation speeds, and an acoustic frequency range approximately 10 times that of the human at both the transmitting and receiving terminals. These factors are indicative of a vocalization capability that is not only highly complex but fundamentally different from that of *Homo sapiens*. The complement of the dolphins' apparent patience with and affinity for human association adds to the challenge for expert professional psychoacoustic and psycholinguistic research on this particular species, as preparation for designing *modi operandi* suitable for treating any encounterable nonhuman codes.

S4. Psychophyiscal Reality of the Distinctive Features of Phonology. DENNIS H. KLATT, *Department of Electrical Engineering and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, Massachusetts.*—Confusions in short-term memory between English CV nonsense syllables are a form of psychophysical data relevant to phonology. A binary distinctive feature is used to divide a stimulus ensemble of phones into two sets. Confusions between phones within a set should be more likely than confusions between phones belonging to opposite sets, otherwise the distinctive feature does not define a categorical perceptual boundary for speakers of the language. Optimal binary features based exclusively on this criterion have been derived, using the 23 consonant confusion matrix data of Wickelgren [J. Acoust. Soc. Am. 39, 388 (1966)]. These optimal features are closely related to the binary distinctive features of phonology.

S5. Phonemic Confusion in Whistled Turkish. JARVIS BASTIAN AND CAROL WALL, *Department of Psychology, University of California, Davis, California*.—A method of whistling the local Turkish dialect has been developed in Northeastern Turkey that involves whistling while executing approximations to the standard articulator maneuvers of the spoken dialect. The signals produced are sufficiently intense and intelligible to those who have learned to use this signaling system to provide an auxiliary means of linguistic communication over long distances in the rugged terrain they inhabit. This system of "whistled speech" is very similar to those that have been recorded in much the same ecological conditions in one of the Canary Islands and in the Pyrenees. However, the Turkish system is based on a far richer phonemic inventory than the others, which are based on Spanish dialects. We therefore analyzed the effectiveness of the Turkish system in transmitting the phonemic contrasts maintained in the spoken language. Confusion test results indicate a severe reduction in the phonemic contrasts transmitted in the whistled version. The nature of these confusions and of their acoustical basis is discussed together with their consequences for sentence intelligibility. [Field work supported by Wenner–Gren Foundation.]

S6. Machine Recognition of Speech. T. B. MARTIN, H. J. ZADELL, R. B. COX, AND M. B. HERSCHER, *Radio Corporation of America*.—This paper summarizes the status of a continuing research program concerned with the recognition of phonemes in both isolated and continuous speech. The acoustic analysis technique employs analog-threshold logic to abstract features that provide the basis for the recognition decisions. These feature-abstraction networks have been organized in a hierarchy of processing levels, ranging from the recognition of broad classes of sounds to decisions on a phoneme-by-phoneme level. As part of this program, a detailed comparison has been made of the characteristics of phonemes in continuous speech as contrasted with well-articulated discrete speech. The development of a continuous speech recognition capability makes possible the realization of several specific applications. Among these are spoken ZIP-code recognition systems that will improve the efficiency of parcel-sorting operation. In this application, it is necessary to recognize continuous strings of digits in a very high-noise background. A speech communications system is also being developed utilizing a speech recognition system at one end of a communications link, a narrow bandwidth channel transmitting the recognition results, and a speech synthesizer at the receiver.

S7. Automatic Speaker Authentication Using Speech Recognition Techniques. W. F. MEEKER, T. B. MARTIN, AND M. B. HERSCHER, *Radio Corporation of America, Camden, New Jersey*, AND DOUGLAS PHYFE AND MARTIN WEINSTOCK, *U. S. Army Electronics Command*.—While some degree of speaker recognition can be achieved using gross measures such as average glottal frequency or average spectrum, it seems certain that, for highly accurate speaker recognition or authentication, it will be necessary to know what is spoken. In applications where there is no control over what the speaker utters, automatic speech recognition becomes a prerequisite for automatic speaker authentication. Considerable success in automatic speech recognition has been achieved using analog-threshold logic and this equipment has been used to select automatically portions of continuous speech for speaker authentication. Characteristics of continuous speech deviate substantially from those predicted by classical phonetics and many of these deviations can be used for speaker authentication. A number of different measures have been studied and automatically extracted. For vowels, the basic measure is the slope of the spectrum. For consonants, spectral characteristics, duration, and sequence of occurrences are useful measures. The over-all accuracy of authentication can be quite high when results for many individual measurements are combined. Results for single and multiple measurements will be discussed. [Work sponsored by the U. S. Army Electronics Command.]

S8. Properties of the Instantaneous Frequency and its Application to Speech Classification. K. HIRAMATSU, D. K. RALEY, R. WACKERBARTH, AND C. L. COATES, *Laboratories of Electronics and Related Science Research, Electrical Engineering Department, The University of Texas, Austin, Texas 78712*.—The instantaneous frequency, corresponding to the derivative of the phase function of the analytic signal, expressed in terms of the zeros of the function and its mean value, is formulated, and the properties of the instantaneous frequency will be illustrated using some typical examples. An expression for the instantaneous frequency in the complex frequency domain has been derived, and has the following properties: (1) When the spectral distribution is symmetric, the average value of the instantaneous frequency corresponds to the mean value of the distribution. (2) When asymmetric, the average value corresponds to the predominating region of the spectrum. Emphasizing or weighting the spectral distribution results in the predomination of frequencies that would not predominate under ordinary circumstances. The average value of the instantaneous frequency then corresponds to that region of the spectrum. The properties of these expressions for the instantaneous frequency will be applied to the classification of a limited number of connected speech sounds; i.e. zero, one, . . ., nine.

S9. Rules for Word Stress Analysis: for Conversion of Print to Synthetic Speech. JANE H. GAITENBY, *Haskins Laboratories, New York, New York*.—Successful conversion of graphemes to English allophones depends on correct assignment of syllable stress (word accent). In turn, stress reflects morphological and syntactic patterns that are intrinsic to the language. Although written English lacks obvious prosodic markers on the word level, its spelling generally represents phonetic realities if stress is not overlooked. (Printed words are also conveniently framed by space.) An important truism is that normal printed texts closely follow spoken grammatical routines. With these facts made operational, and coupled to a modest storage of letter sequences (with appropriate syllables treated as stable stress elements), a fair degree of prosodic contour prediction is attainable. In this paper, preliminary rules for deriving stress information from a printed input are exemplified. The usefulness of these rules in a program for basic intonation in a synthetic speech output is demonstrated. [This research on reading machines for the blind was made under contract to the Research and Development Division of the Prosthetic and Sensory Aids Service, Veterans Administration, New York.]

S10. Speaker-Listener Dialect Error in Discrimination Testing. CORNELIS W. KOUTSTAAL AND MARTHA HOSACK, *Department of Speech, Bowling Green University, Bowling Green, Ohio 43402*.—Articulation curves were obtained from a group of listeners who listened to speech discrimination-test words in their own dialect and a dialect different from their own. The speakers were from the General American dialect and the Southern American dialect regions. A male and a female speaker represented each dialect. The listeners were all from the General American dialect. The curves were derived from the mean values of 20 listeners for each of the nine sensation levels. Each listener responded to the test stimuli only at one sensation level, and to both dialects. The responses to the female speech sample showed a consistent difference in favor of the listener's own dialect. The responses to the male speech sample did not yield consistent differences in favor of either dialect. The difference between the responses to the male and the female speech sample is discussed.

S11. Syntactical versus Temporal Cues for Speaker Switching in Natural Dialogue. LOUIS J. GERSTMAN, *Queens College, C.U.N.Y., New York*, STANLEY FELDSTEIN (nonmember), AND JOSEPH JAFFE (nonmember), *The William Alanson White Institute and Columbia University, New York City.*—Intercomparison of two data domains permits an assessment of the relative importance of pauses and syntax as determinants of speaker switching in spontaneous dialogue. The pause and switching data derive from the output of the automatic vocal transaction analyzer, which samples the presence and absence of each participant's speech every 300 msec. The syntactic data derive from boundary markings (points of linguistically permissible phrase endings) on 10 642 words of text transcribed from four 15-min dialogues. Synchronization of both data bases revealed that switching occurs nine times more often after pauses than after nonpauses, and 21 times more often after boundaries than after nonboundaries. The two effects are nonadditive, since the co-occurrence of both cues makes switching 42 times more likely than the co-occurrence of neither. When in conflict, the syntactic cue outweighs the pause cue, since switching occurs 2.6 times more often after a nonpausal boundary than after a nonboundary word followed by a pause. [Research supported by grants from the National Institute of Mental Health.]

WEDNESDAY, 15 NOVEMBER 1967 — BURGUNDY EAST, 2:00 P.M.

Session T: Underwater Acoustics IV: Ocean Boundaries

BURTON G. HURDLE, *Chairman*

Invited Papers (25 minutes)

T1. Measurement of the Roughness of the Ocean Bottom. C. S. CLAY, *Hudson Laboratories of Columbia University, Dobbs Ferry, New York, 10522.*—We would like to consider a description of the ocean floor that is particularly useful in ocean acoustics. The earth's visible surface is described by means of topographic maps and aerial photographs. To some extent, attempts to describe the ocean floor have followed this practice. However, the problem of constructing topographic maps is immense, and deep-sea photographs yield a postage-stamp view of the bottom. If we had long-range sonographs, most of our problem would be solved. (Side-looking sonars are a step in this direction.) Marine geologists have classified the ocean floor into physiographic provinces, and within each province the floor has similar properties. Experimental data have shown that in addition to the acoustic properties of the floor materials, the roughness of the interface is extremely important. In theoretical studies, roughness parameters such as the rms height of irregularities and their correlation functions over the illuminated area can be used to calculate the reflected and scattered signals. These are also measurable quantities. We urge those who study the acoustic properties of the bottom to report their results in standard physical units and statistical properties of the roughness. The development of an empirical bottom-roughness scale like the old sea-state scales and reflection-loss scales, etc., should be avoided. [Hudson Laboratories of Columbia University Informal Documentation No. 139. This work was supported by the U. S. Office of Naval Research.]

Contributed Papers (10 minutes)

T2. The Variability of Bottom Reflected Signals Using the Deep Research Vehicle ALVIN. LAURENCE C. BREAKER AND ROBERT S. WINOKUR, *U. S. Naval Oceanographic Office, Washington, D. C. 20390.*—Two dives were made using the deep research vehicle ALVIN to investigate the variability of normal incidence bottom reflected signals at 12 kHz. The measurements were made over a locally smooth bottom in a water depth of about 4700 ft in the tongue of the ocean. Approximately 800 bottom reflected signals were recorded while ALVIN hovered at three levels above the ocean bottom; 500, 2000, and 4500 ft. For each depth, frequency distributions, atuocorrelation coefficients, and coefficients of variation were computed using both relative peak amplitude and energy levels. The results indicate that significant changes in bottom reflectivity can occur in a short duration and that the observed variations are probably cuased by changes in the reflective characteristics of the ocean bottom as the vehicle drifted in a horizontal direction. Changes in the bottom are suggested by the data collected at the deepest level, which exhibits a pronounced shift in signal level and a bimodal frequency distribution of pulse amplitudes. In addition, the observed fluctuations generally increase with increasing distance above the bottom.

T3. Amplitude Stability of Bottom Bounce Sound Propagation in the Hawaiian Area. F. H. FISHER, *University of California, San Diego, Marine Physical Laboratory of the Scripps Institution of Oceanography, San Diego, California 92152.*—Measurements of the coefficient of variation have been made of short (5 msec) 5.5 kHz pulses at a range of 5.5 mm in about 2500-fathom depth at position 18°58′ N and 156°16′ W. Pulses received at hydrophones mounted on FLIP at 300-ft depth show the bottom-bounce path to be highly stable compared to the waterborne. Photographs of the received pulse structure as well as numerical data will be presented.

T4. The Reflectivity of Two Bottom Areas as Measured by Total Energy Reflection Coefficients. FRANCIS R. MENOTTI, *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—In a paper presented at the 72nd meeting of the Acoustical Society of America, the author discussed the high variability in the peak amplitude bottom reflection coefficients obtained using pulsed CW signals over an abyssal plain area. These results led to an investigation into the use of a reflected-to-incident energy ratio as a more suitable description of bottom reflectivity. Ratios between the integral-square values of the bottom-reflected and direct-path signal envelopes were computed for 100 msec pulsed CW and 250-msec linear frequency modulated (LFM) signals. The frequencies were 1080 and 3700 Hz for the CW; the LFM bands were centered on these frequencies. Results indicate that the integral-square ratios, which are essentially energy ratios, are as variable as the peak reflection coefficients. The variability, expressed as the interval bounded by the mean plus and minus one sigma, is higher for the 3700-Hz data. For both frequencies and signal types, this variability often exceeds 4 dB over survey areas only a few miles in extent.

T5. Coherence of Ocean Bottom Reflected Acoustic Pulses. SALVATORE R. SANTANIELLO, *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—Experiments have been conducted at abyssal plain locations in the Atlantic Ocean with linear frequency modulated pulses 250 msec in length. The frequencies were 1100 Hz (±50 Hz) and 3700 Hz (±300 Hz). Nonoverlapping direct and bottom-reflected instantaneous signals received at two vertically spaced (1100 ft. apart) hydrophones were polarity correlated. Cross-correlations were also obtained (1) between a group of consecutive reflected arrivals at one hydrophone using one arrival as replica and (2) between arrivals at the two hydrophones. These temporal and spatial coherence measurements resulted in mean peak correlation values >0.5 as a function of grazing angle and frequency. At one location, 50 consecutive reflected pulses were consistent in their modulation characteristics and were analyzed for different time-bandwidth products. The higher-frequency 600-Hz bandwidth data resulted in a lower mean peak correlation value because of the multipaths within the reflected pulses caused by the fine grain structure of the bottom. A reduced bandwidth (300 Hz) resulted in a larger spread in peak correlation values, but these values had a greater average and produced less complicated correlation functions. By use of spectral analysis, these stationary reflected pulses have also provided a means for inferring the apparent acoustic complex structure of ocean bottoms.

T6. Ocean Surface Reverberation Mechanisms and Their Laboratory Simulation. H. S. HAYRE, *University of Houston, Houston, Texas.*—Ocean surface roughness is generated by a combination of gravity and wind driven waves. These water/air interface perturbations are directly related to water motion just below this interface. This circular motion of the water may extend down to a depth of up to approximately two times the trough to crest measure of such waves. The ocean reverberation analyses to date have employed an interface description in the form of wave height measurements taken at a point and/or its associated spectrum, with very reasonable results. The experimental reverberation data has not been fully exploited to determine the significance of the said underlying water motion as the major contributor to the acoustic signal intensity scattered into the water body. It is this aspect that is investigated both theoretically with a model of the moving surface as well as the water mass underneath it, and experimentally in a laboratory tank with controlled and repeatable conditions using 1 Mc/sec carrier pulse of widths 4–6 μsec. The water motion was simulated by forced flow of water just underneath the air/water intersurface. The position of a small specular reflecting test plate was varied vertically between the transmit-receive transducer and the air/water interface in order to identify and separate subsurface volume and water/air interface reverberations. Various mechanisms generating the frequency spread of the acoustic signal incident on the surface are also discussed. [This work is sponsored by the Office of Naval Research, under Contract.]

T7. Backscattering of Underwater Sound from a Wind-Driven Model Sea Surface. HERMAN MEDWIN, ERNEST C. BALL, AND JOHN A. CARLSON, *Naval Postgraduate School, Monterey, California.*—Model seas have been generated by adjustable winds over a large anechoic tank. The distribution of heights and slopes and the wave height energy spectrum have been determined for a homogeneous section of this statistically-stationary, anisotropic surface. This known, nearly-Gaussian "sea" has been used to predict the backscattered sound according to several different theoretical "solutions". The predictions are compared with many thousands of instantaneous values of scattered pressure for sound incidence θ_i from normal to 80° and surface roughnesses $0.01 < g < 60$ where $(g)^{\frac{1}{2}} = 2k\sigma \cos\theta$, with σ = rms height. Measured mean-square pressures show particularly good agreement with theory when the effect of diffraction is added to the surface spectrum description as developed by Eckart (1953), Marsh (1963), and Wetzel (1966) for $g<1$. There is decidedly poorer support for the various statistical approaches used by Eckart, Isakovitch, Cox, and Beckmann in the case of rough surface scatter $g \gg 1$. The distribution of amplitudes is found to be Rayleigh, except near normal incidence, where it becomes Gaussian as the roughness decreases to $g \ll 1$. [Work supported by Naval Ship Systems Command.]

T8. Frequency Spectra of Forward Scattered Underwater Sound from a Traveling Sinusoidal Surface. WILLIAM I. RODERICK, *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—An experiment has been conducted in a model tank to determine the frequency spectrum of an acoustic signal reflected from a traveling sinusoidal surface. An acoustic signal of 1.85 MHz was transmitted continuously and projected toward the surface (beamwidth 3.5°). A vibration generator produced water gravity waves; wavelength and wave-height measurements were made at the point of reflection. The forward scattered signal was envelope detected and analyzed for spectral content. Frequency information regarding the carrier and sidebands was also obtained. A scattering integral developed in the literature was evaluated for a traveling sinusoidal surface to compare the experimental results. Analytical predictions were made, based on the concept that the surface waves phase modulate the acoustic signal. Important observations are: (1) both amplitude and phase modulation are present, (2) the frequency spectrum is symmetrical about the carrier and can contain only sideband frequencies that are multiples of the surface frequency, i.e., $f \pm nf_s$ (where f = acoustic frequency, $n = 0, 1, 2, 3, \cdots, f_s$ = surface frequency), (3) the amplitudes of the frequency components are dependent on transmitting and receiving geometry and measured surface characteristics.

T9. Surface Reflection Modulation of an Underwater Explosion Spectrum. ERMINE A. CHRISTIAN, *U. S. Naval Ordnance Laboratory, White Oak, Silver Spring, Maryland 20910.* – For a simple harmonic source near the surface of the ocean, the interference pattern of direct and surface-reflected waves is described by the well-known Lloyd mirror dependence. With a pulse source, such as an underwater explosion, the interference effect is a modulation of the original pulse spectrum by a factor that depends upon frequency and the space variables. When the source and the receiving hydrophone remain at constant (though not necessarily the same) depths, both the amplitude and the period of the modulation term increase with increasing range. Depending upon the burst-receiver geometry and the reflectivity of the surface, at some frequencies the spectral energy level of the direct wave might, theoretically, be increased by as much as a factor of 4. In this paper, predicted differences between the original and the surface-modulated pulse spectrum are shown for several geometries and reflectivity coefficients, and are compared with measurements from small underwater explosions. [This work was supported by the Advanced Research Projects Agency.]

T10. Effect of Sediment Shear Waves on Bottom Reflection Losses. HALCYON E. MORRIS, *Naval Undersea Warfare Center, San Diego Division, San Diego, California 92152.*—The question of whether bottom reflection losses calculated with the viscoelastic model for bottom sediments are significantly different from the results from the simpler complex velocity model has been of recent interest. The physical difference in the two models is the generation of shear waves in the viscoelastic model as a result of sediment rigidity and viscosity defined by complex Lamé constants. Results are presented where the same density, compressional velocity and attenuation in the sediment were used in both models. Varying amounts of rigidity were introduced into the sediment layer. Consideration of the shear waves in the viscoelastic model generally results with higher bottom reflection losses than the complex velocity model. The difference is most notable when the sound velocity in water and the compressional velocity in the sediment are nearly the same.

T11. Influence of Hollow Shells on the Porosity-Sound-Velocity Relationship in Some Marine Sediments. JAMES J. GALLAGHER, *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—Representative clean globigerina tests were constructed from available data, and internal void volumes were computed. For a median grain size of 0.5 mm, a value of 45% was assigned to the interstitial porosity. The effect of adding a computed fluid-filled internal porosity of about 80% is to produce a mean bulk porosity of 60%, or a change of 15% from that of a solid grain aggregate. If the packing of solid and hollow particles of the same uniform size and shape is similar, the degree of grain-to-grain contact and of interstitial porosity is similar. Therefore, sound velocity through these uniform aggregates should be of comparable value. However, the addition of internal porosities of the hollow shells offsets the porosity versus sound-velocity relationship. Porosity values for the high-carbonate hollow shells should be reduced by 15%–20% to fit the sound velocity versus interstitial porosity curve. [Data obtained from the Marine Geophysical Survey substantiate the results of these preliminary computations.]

T12. Backscattering of Sound from the Ocean Surface. M. VERTNER BROWN AND R. ALFRED SAENGER, *Hudson Laboratories of Columbia University, Dobbs Ferry, New York, 10522.*—The backscattering of sound from the ocean surface has been measured in the frequency range 60 to 4000 Hz. Shots were fired at a depth of 3000 ft and "simultaneously" received on hydrophones directly above at 3000, 2000, 1000, and 100 ft. Concurrently surface roughness measurements were taken using a gyrostabilized accelerometer buoy. As a general rule, all scattering coefficients at a given frequency fall on the same curve when plotted against $(\sin\phi_i+\sin\phi_s)$. ϕ_i and ϕ_s are, respectively, the grazing angles of incidence and backscattering. (Hudson Laboratories of Columbia University Informal Documentation No. 136. Work supported by the U. S. Office of Naval Research.)

THURSDAY, 16 NOVEMBER 1967 — BURGUNDY EAST, 9:00 A.M.

Session U. Ultrasonic Visualization V; Underwater Viewing and Material Inspection

JOHN V. BOUYOUCOS, *Chairman*

Invited Papers (25 minutes)

U1. Information Rate, Resolving Power, and High-Resolution Sonar Systems. D. H. BROWN, *U. S. Navy Mine Defense Laboratory, Panama City, Florida.*—Ultrasonic visualization systems with very high resolution can best be described using concepts similar to those developed in the study of optical imaging systems. Such systems are thus quite naturally called acoustic imaging systems. The concepts of information rate, measured in resolution elements per radian second and resolving power, which is related to the spatial frequency bandwidth of a system, are developed and used as a basis for a historical review of some of the techniques that have been applied to underwater acoustic imaging. Unfortunately, a common characteristic of most systems developed in the past is their limited success. By considering designs that will simultaneously offer high resolving power and information rate, some new techniques that may result in more successful underwater acoustic imaging systems are introduced and discussed.

U2. Ultrasonic Image Systems for Nondestructive Testing. HAROLD BERGER (nonmember), *Argonne National Laboratory, Argonne, Illinois.*—Ultrasonic imaging techniques are finding increased application in the medical and industrial fields because the image presentation often offers the advantage of improved interpretation capability for the ultrasonic test. In the nondestructive testing field, ultrasonic imaging inspection methods have recently been expanded to include television techniques; the operation and application of these techniques are emphasized in this report. In addition to interpretation advantages, television methods can also present improved resolution capabilities and faster inspection speeds, as compared to other ultrasonic inspection methods. Continuous-wave ultrasonic television systems have shown promise as inspection methods for relatively thin, flat material in which inspection is desired to determine such things as bonding, particle size variations, or homogeneity. Flat, reactor fuel plates and thin metal spot welds are examples of materials for which this type of inspection has been shown to be useful. Pulsed inspection systems have also been demonstrated, with relatively slow image presentation rates. These systems, when further developed, offer promise for greater range of application. [This work was performed under the auspices of the U. S. Atomic Energy Commission.]

U3. Possibility and Limitations in Sonoholography. P. GREGUSS (nonmember), *Railway Scientific Research Institute, Budapest, Hungary.*—The geometric similarity of sound and light waves has simulated numerous attempts at visible sound images. Since 1964, efforts have been made to produce and to reconstruct holograms by ultrasonic waves. Sometimes there is a chance to get information from a hologram without wavefront reconstruction by using Moire fringes techniques. Wavefront reconstruction by ultrasonic waves promises a new way to overcome the difficulties issuing from the wavelength differences. Using a new acoustical–optical lens system, ultrasonoholograms can be obtained that can be treated as an ordinary laser hologram. Ultrasonoholography has its future not only in the nondestructive testing of materials, but also in ultrasonic diagnostics. Possibilities and limitations will be discussed.

U4. An Underwater Acoustic "Television" System. R. W. G. HASLETT, *Kelvin Hughes, a Division of Smiths Industries, Ilford, Essex, United Kingdom.*—It may not be widely known that, over the period 1949–1954, an experimental model of a closed-circuit underwater acoustic "television" system for use in muddy water, was developed in the United Kingdom under an Admiralty contract. The object to be viewed was irradiated with ultrasonic waves and a real acoustic image of the object was formed by a concave mirror. A special converter tube, rather similar to a television camera, transformed the image to a television picture. Two programs of work were pursued side by side: (a) the development of the image converter tube and electronic circuits, and (b) the design of the mirror assembly. Subsequently, the whole system was subjected to detailed investigation in a water tank and at sea, to determine the resolution, the maximum range obtainable and the appearances of a number of objects having simple shapes.

Contributed Papers (12 minutes)

U5. Experimental Ultrasonic Image System for Underwater Use. G. L. SACKMAN, A. F. BARTA (nonmember), G. C. CASWELL (nonmember), AND K. G. ROBINSON (nonmember), *Naval Postgraduate School, Monterey, California 93940.*—An underwater object is insonified by a 455-kHz pulse from a separate projector, and the reflected waves are focused onto a 9×9-element mosaic transducer fabricated from a single ceramic block, located at the focal plane of 90-cm-diam parabolic reflector. The transducer voltages are amplified by a set of 81 identical amplifiers with range gates, then rectified to charge storage capacitors. The capacitors are scanned in sequence by solid state switches driven by a microcircuit logic network, which also generates sweep voltages for the intensity modulated display. Resolution of $\frac{1}{2}°$ and tangential sensitivity of -110 dB *re* 1 W/cm^2 have been measured, with interchannel crosstalk on the order of -20 dB. Digital-computer simulation of the system has emphasized the importance of aperture shading and proper reconstruction of the sampled image, in improving image quality. [This work was supported in part by the Office of Naval Research.]

U6. Utilization of the Liquid Surface Levitation Effect as a Means of Ultrasonic Image Conversion for Materials Inspection. OTTO R. GERICKE AND ROBERT C. GRUBINSKAS (nonmember), *U. S. Army Materials Research Agency, Watertown, Massachusetts 02171.*—An ultrasonic image conversion cell utilizing the liquid surface levitation effect has been developed. This effect is based upon the principle that the surface of a free liquid will be deformed in accordance with the cross-sectional intensity distribution of an impinging ultrasonic beam. The image conversion cell has been successfully applied for nondestructive testing purposes. It can be coupled directly to an ultrasonically excited test specimen to obtain characteristic liquid-surface relief patterns depicting internal discontinuities. A significant advantage of the conversion cell is the fact that it does not require the test specimen itself to be immersed in a liquid. Also, because the conversion cell produces a "real time" image, pulsed, frequency-modulated, and continouus-wave modes of ultrasonic excitation can be used, and thus, various types of ultrasonic images can be obtained.

U7. Image Scanner Using Diode Switching. E. E. SUCKLING (nonmember) *Downstate Medical Center, Brooklyn, New York* AND J. R. HENDRICKSON (nonmember), *Oceanic Institute, Hawaii.*—We have used in the past, as an ultrasonic image conversion system, a capacity probe tracing mechanically over the dry face of a crystal set in a tank wall. An image took several minutes to scan. The present system comprises 64 electrodes fixed in a line 3 in. long across a crystal. Diode switching brings in the electrodes consecutively. A scan of the line takes 1/300 sec. Experiments with a rotating sonic mirror indicate that a sonic image can be moved across the line of probes in $\frac{1}{5}$ sec. The system then gives a video signal allowing the construction of a converted sonic picture comprising 4000 picture elements 5 times each second. Noise levels depend mainly on the switching diodes and since band widths can be quite restricted, are expected to be in the tens of microvolts. Illustrations of the video signal obtained from a parallel beam of 3 Mc/sec continuous wave ultrasound focused onto the array and from an image of two slots $\frac{1}{16}$ in. wide and spaced $\frac{1}{8}$ in. will be shown.

U8. Evaluation of the Pohlman Technique—a Historical Note. H. E. VAN VALKENBURG, *Sperry Products Division, Automation Industries Inc., Danbury, Connecticut.*—A brief report on a program sponsored by the Department of the Navy in the late 1940's to evaluate the Pohlman acoustic imaging technique for nondestructive testing. Reports were issued under the contract (NOas 8868), but no information was published. The results obtained, some very impressive, and the basic limitations encountered, are of current interest with respect to visualization technology. The system comprised a small water tank having a 20-in. beam path, transducer, sample holder, plastic lens, Pohlman window, and rf power source. Transducers were 1.0-in. diam quartz of 1 to 10 MHz frequency. The rf source was frequency swept to minimize standing waves. Best results were obtained on thin, flat, metal specimens; poorest on contoured samples. The technique appeared to be limited to the through-transmission method of inspection. Photographs of the equipment and of the images obtained on a variety of specimens will be shown.

THURSDAY, 16 NOVEMBER 1967 — SILVER CHIMES EAST, 9:00 A.M.

Session V. Engineering Acoustics II: Acoustic Ranges and Facilities

ROBERT J. BOBBER, *Chairman*

Invited Papers (20 minutes)

V1. Atlantic Undersea Test and Evaluation Center (AUTEC). L. L. JACKSON, JR. (nonmember), *Atlantic Undersea Test and Evaluation Center, International Airport, West Palm Beach, Florida 33406.*—The Navy's Atlantic Undersea Test and Evaluation Center in the Bahama Islands is one of the largest facilities in the world in which acoustics plays a major role as both a measurement variable and measurement tool. The aims, activities, and purposes of AUTEC will be presented in a 20-min film that will serve as an introduction to the three following papers on the weapons, acoustic, and sonar ranges.

V2. AUTEC Weapons Range. WILLIAM ONEBY (nonmember), *Atlantic Undersea Test and Evaluation Center, International Airport, West Palm Beach, Florida 33406.*—The AUTEC Weapons Range is a three-dimensional tracking facility instrumented to track both in-air and in-water targets. The range presently consists of a 35-mile-long area of the Tongue of the Ocean. The in-air tracking instrumentation is located on shore. The in-water tracking instrumentation includes large hydrophone arrays designed for deep-water operation. The acoustic instrumentation and tracking technique are described.

V3. AUTEC Acoustic Measurement Range. R. B. ALSPAUGH (nonmember), *The Atlantic Undersea Test and Evaluation Center (AUTEC), International Airport, West Palm Beach, Florida 33406.*—The AUTEC Acoustic Range is a 5-by-20-mile facility for measuring and analyzing noise radiated from surface ships or submarines. The instrumentation system is discussed in terms of data acquisition and processing, preliminary analysis system, communications and navigation and underwater instrumentation. The bottom-moored hydrophone array will become operational by December 1967.

V4. AUTEC Sonar Calibration Range. F. P. HERRING (nonmember), *The Atlantic Undersea Test and Evaluation Center (AUTEC), International Airport, West Palm Beach, Florida 33406.*—The AUTEC Sonal Calibration Range will be developed in three phases. Phase I will provide sonar hearing alignment and a range calibration. Phase II and Phase III will provide on-site sonar directivity pattern measurements in receiving and transmitting modes, source-level calibration, etc., for surface ships and submarines. This paper presents environmental and electroacoustical characteristics of the range.

Contributed Papers (10 minutes)

V5. Calibration of a Long-Base-Line Underwater Acoustic Tracking Range. MILES W. MCLENNAN, *AC Electronics Defense Research Laboratories Santa Barbara, California.*—In 1967, AC-DRL completed its Santa Cruz Acoustic Range Facility (acronymed SCARF) in the Santa Cruz Basin off Southern California. The range consists of four broad-band hydrophones set in a 6000-ft rhombus in 4000 ft of water. Cables run to signal processing electronics and a digital computer at a fixed site on Santa Cruz Island. The range is intended for precision multitarget underwater three-dimensional tracking and noise measurement. This paper describes the acoustic calibration experiment that determined the separation and depths of all hydrophones to less than a foot. A surface ship with fixed pinger was positioned at points in a 10 000-ft-square grid centering on the array, and 175 acoustic pings were emitted and recorded by the signal processors and computer. After the run, the computer did a least-squares fit of the hydrophone and pinger locations to the recorded transit times, using local tides and ray bending and transit time corrections form sound velocity profiles taken during the experiment. Using an edited set of 166 pings, the solution residual was 0.68 ft.

V6. Directional 420-Hz Sound Source. MORTON KRONENGOLD, *University of Miami, Institute of Marine Sciences, Miami, Florida 33149* AND WILLIAM TOULIS, *North American Aviation, Inc., 5701 West Imperial Highway, Los Angeles, California 90009.*—A 24-ft compliant-tube parabolic reflector, with a flextensional transducer at its focus, has yielded acoustic source levels up to 122 dB/bar/yd at 420 Hz in 85 ft of water off Fowey Rocks near Miami, Florida. The projector is part of an ONR supported research program to study underwater sound propagation and related environmental parameters. Modulation signals of continuous-wave pulses, and pseudorandom sequences are transmitted by wire from the Institute of Marine Sciences Laboratory to the transducer, a distance of 12 miles. Bottom-mounted hydrophones are employed at various distances, with the principal sensors on the island of Bimini which lies 45 miles eastward across the Florida Straits. The reflector provides a mainlobe of 28° at the half-power points; the back lobes are down more than 25 dB and the directivity index is 15 dB. The effective mechanical Q of the transducer–reflector combination is 4. At the present depth, the transducer is cavitation limited to an acoustic output of 4 kW.

V7. A Probabilistic Model for Acoustic Sound Ranging. ROBERT P. LEE (nonmember), *Atmospheric Sciences Laboratory, U. S. Army Electronics Command, White Sands Missile Range, New Mexico.*—A probablistic model for acoustic sound ranging in a nonhomogeneous atmosphere has been developed. The model is based on a linear array of microphones. Error contour curves are presented that characterize the effects of of the nonhomogeneous atmosphere upon sound ranging results.

V8. Sonar Transducer Calibration in a Controlled Pressure–Temperature Environment. C. E. GREEN AND J. R. ROSHON (nonmember), *Naval Undersea Warfare Center, San Diego, California 92152.*—It is known that both temperature and pressure affect the performance of most sonar transducers. A system has been installed at the Transdec facility of the Undersea Warfare Center, San Diego Division, so that a transducer can be subjected to controlled temperature from 40° to 90° F and/or pressure from 0 to 2000 psi during calibration. The transducer to be calibrated is mounted inside an acoustically transparent vessel 3 ft in diameter by 6 ft long. Water is circulated through the vessel and through a deck-mounted refrigerating-heating unit. A standard transducer known by other means to be independent of temperature and pressure shows no pattern or response deterioration when checked in the vessel. Examples will be presented that demonstrate the changes in performance in other transducer due to variations in temperature and pressure.

THURSDAY, 16 NOVEMBER 1967 MEDALLION EAST 9:00 A.M.

Session W: Psychological and Physiological Acoustics V: Complex Physiological Processes

PETER J. DALLOS, *Chairman*

Contributed Papers (10 minutes)

W1. Acoustic Vestibular Stimulation in the Guinea Pig. DONALD E. PARKER AND MILLARD RESCHKE (nonmember), *Miami University, Oxford, Ohio.*—Observations of guinea pig eye movements associated with periodic (AC) and static (DC) pressure variations in the external auditory meatus are presented. Eye movements that are associated with normal vestibular stimulation were studied. These eye movements were determined by direct observation with a microscope and by electronystagmography. Stimulus intensities ranged up to 7 in. Hg. AC pressure variations were in the range of 0.25–10 Hz. DC pressure shifts were maintained for periods up to 15 sec. These stimuli produced three general types of eye movements: oscillatory eye movements, "counterrolling," and nystagmus. Oscillatory eye movements at the same frequency as the stimulus, were observed in conjunction with AC stimuli at intensities of 2–3-in. Hg. Increasing stimulus intensity to approximately 5-in. Hg produced a movement analogous to that seen in normal counterrolling and, in some cases, nystagmus. DC pressure shifts in the range of 2–3-in. Hg produced counterrolling. With DC pressures of 3–4-in. Hg nystagmus was generally observed in conjunction with the counterrolling. These observations provide support for the hypothesis that acoustic stimuli may produce activity in vestibular receptors.

W2. Visual Tracking of Auditory Stimuli. R. W. STREAM, E. T. WHITSON (nonmember), AND V. HONRUBIA, *Bill Wilkerson Hearing and Speech Center and the Division of Otolaryngology, Vanderbilt University School of Medicine, Nashville, Tennessee.*—Inside an anechoic chamber electro-occulographic techniques were used to record the eye movements in 20 subjects who were requested to track with their eyes a steady and continuously moving auditory stimulus. A motor-driven rotary switch made successive contact with 24 loudspeakers mounted side by side in an arc 180° wide and 11 ft in diameter. At angular sound velocities ranging from 15°–180°/sec, the eye movements were recorded under conditions of light and darkness. As the angular velocity of the stimulus increased, the horizontal displacement of the eyes decreased from an average maximum amplitude of 90°–60°. At low velocities, the eye movements followed precisely the position of the sound source. At high velocities (i.e. larger than 130°/sec) mostly saccadic movements were produced. The tracking ability of the eye, judged by the qualitative analysis of the recorded movements, was better in tests performed in the light rather than in darkness.

W3. Human Equilibrium During Acoustic Stimulation by Discrete Frequencies. C. STANLEY HARRIS (nonmember) AND HENRY C. SOMMER, *Aerospace Medical Research Laboratories, Wright-Patterson Air Force Base, Dayton, Ohio.*—Previous research has demonstrated that high-intensity broad-band noise has adverse effects on human equilibrium. In the present study, an attempt was made to determine whether frequency (Hz) of acoustical stimulation is an important variable in the ability of subjects to balance on narrow rails. Two groups of subjects performed the rail task during exposure to pure tones of 100, 260, 590, 1500, 2500 Hz, and a control condition. One group was presented the tones with equal intensity to both ears (symmetrical exposure) and the other group was presented a higher intensity to the left ear than to the right ear (asymmetrical exposure). A decrement in rail-task performance was obtained only at 590 Hz for the symmetrical exposure group and only at 1500 Hz for the asymmetrical exposure group. The decrement obtained with symmetrical exposure was less than the decrement for asymmetrical exposure. This result supports previous findings that asymmetrical exposure has more adverse effects on human equilibrium than symmetrical exposure. Both frequencies at which decrements were obtained have been found in previous studies to have the lowest thresholds for vestibular stimulation as determined by a slight shifting of the visual field in normal hearing subjects and nystagmus measures in deaf individuals. The results are interpreted as a possible demonstration of the direct effects of noise on the vestibular system.

W4. Echolocation: Resolution of Target Range by Bats. JAMES A. SIMMONS, *Auditory Research Laboratories, Princeton University, Princeton, New Jersey 08540.*—Echolocating bats (*Eptesicus fuscus*) were trained to discriminate between identical targets located at different distances from the bat. By means of a simultaneous discrimination procedure, the threshold for detection of differences in target range was determined. At an over-all range of 30 cm, the minimum detectable difference between the nearer and farther targets was 11 mm. The bats were blinded, and they emitted characteristic echolocation cries throughout the discrimination process. Three cues are available to the bats for detecting the difference in target range: the apparent size difference between the targets, the longer travel time for the echo from the farther target than for the nearer target, and the instantaneous frequency differences between the returning FM echoes slightly offset in time. Concurrent experiments indicate that *Eptesicus* probably measures target range by echo time delay.

W5. Signal Characteristics of the Calls in the Bullfrog's Vocal Repertoire. ROBERT R. CAPRANICA, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—The vocal repertoire of colonies of bullfrogs maintained in a seminatural environment in the laboratory consisted of a distinct set of stereotyped calls: (1) mating calls made only by males; (2) three types of calls involving territorial behavior—one type made only by males, one type made only by females, and a third type made by both males and females; (3) release calls made by both sexes; (4) warning calls made by both sexes; and (5) distress calls made by both sexes. Each call is characterized by a different gross temporal pattern. The periodicity of the temporal fine structure is different for each type of call, with the mating call having the highest periodicity (100/sec). The spectral distribution of energy in each call is concentrated in the range 200–2000 Hz. The bullfrog's auditory system is well suited to detect and encode the temporal and spectral features of the acoustic waveforms in these calls.

W6. Auditory-Nerve Responses as a Function of Repetition Rate and Background Noise. DONALD C. TEAS, *Communication Science Laboratory and Department of Psychology, University of Florida, Gainesville* AND GRETCHEN B. HENRY, *Bioacoustics Laboratory, Eye and Ear Hospital, University of Pittsburgh.*—Auditory nerve responses to 2- and 6-kHz tone pips were analyzed for changes in magnitude and latency. The signals were presented at repetition rates from 2/sec to 160/sec with and without background noise present. The bandwidths of the noises and signals were identical, although unlike signal and noise were also paired. The responses to the 2-kHz pip presented at 2/sec in weak 2-kHz background noise showed an increase in magnitude as compared to the no-noise condition. The responses to the 6-kHz pip showed a similar increase in magnitude for increases in repetition rate up to 10/sec. However, at 10/sec, 6-kHz background noise was maximally effective (over-all repetition rates) in reducing response magnitude. Associated with response enhancement was a decrease in peak latency. Stronger noise and faster repetition rates increased the latency and decreased response magnitude. [This research was partially supported by PHS grant.]

W7. Neurological Activity During Threshold Tracking in the Cat. JAMES C. SAUNDERS, *Auditory Research Laboratory, Princeton University, Princeton, New Jersey*—Cats were trained to track auditory thresholds to a 1.0- and 2.0-kHz tone, acoustic clicks, and electrical pulses delivered to the cochlear nucleus. The frequency-synchronized discharges produced by the tonal stimuli were monitored at the cochlear nucleus with a wave analyzer as the animal reached its behavioral threshold. A direct empirical relationship was observed between the behavioral threshold and the amplitude of neural activeity detected at the cochlear nucleus. Evoked potential discharges were monitored at the auditory cortex as experimental animals tracked a behavioral threshold to transient stimuli. The results suggest that the absence of cortical activity is highly correlated with the inability to perceive the stimulus. The relationship between neural activity and the behavioral threshold indicate that the amplitude of neural activity is directly related to the perception of loudness.

W8. Olivocochlear Bundle Regulation of Spontaneous and Tone-Evoked Activities of Single Units in Cochlear Nucleus. A. STARR AND J. WERNICK, *Stanford University, Palo Alto, California.*—132 units were recorded extracellularly in 19 decerebrate unanesthetized cats paralyzed with Flaxedil. Tetanic electrical olivocochlear bundle stimulation at the facial genu (20–200-μA, 0.1-msec pulses, 200/sec) decreased spontaneous discharge rates in 30% of the units, increased it in 18% and had no effect in 52%. Effects of olivocochlear bundle stimulation on tone-evoked responses (Tone bursts, 200 msec in duration, rise and fall times of 25 msec) were analyzed with the units' "best" frequency presented at different intensities. Three types of response modifications occurred. (1) Decrease in discharge rate equivalent to a 5–10-dB shift in the intensity function; (2) increase in discharge rate, and (3) either a decrease or increase in firing rate as a function of tone intensity. The unit's threshold presence of inhibitory surround, and the type of intensity function (montonic or nonmontonic) were correlated with these response modifications. Experiments on animals with cochleas acutely destroyed showed olivocochlear bundle stimulation to still be effective in modifying spontaneous discharges in cochlear nucleus. We conclude that olivocochlear bundle has a direct effect on cochlear nucleus in addition to its well-known receptor effects.

W9. Olivocochlear Bundle: Relationships to Signal Discrimination in Noise. JAMES H. DEWSON, III, *Stanford University School of Medicine, Palo Alto, California.*—Monkeys, trained to discriminate between human speech sounds presented at 70 dB SPL in 2400-Hz low-pass noise of different intensities, were found significantly impaired in their performance following surgical section of the crossed OCB. There is absolutely no loss in the ability to discriminate between the signals in the absence of noise. The deficit is related to "perceptual" (as opposed to physical) signal-to-noise ratio: high intensity noise *per se* is insufficient to cause performance decrement if its passband provides inadequate masking of the speech stimuli (e.g., 2400 Hz high pass). The magnitude of the postoperative deficit is apparently related to the extent of destruction of the fibers of the OCB.

W10. Differential Effects of Cortical Lesions upon Various Types of Auditory Discriminations in the Monkey. D. N. ELLIOTT AND L. A. FRAZIER (nonmember), *Wayne State University and Henry Ford Hospital.*—Following preablation training on frequency, intensity, duration, and complex spectral difference discrimination tasks at three different frequencies, postablation performance was determined. Variations in the testing procedure were also introduced to determine the extent to which changes in discrimination performance reflected the specific nature of the discrimination task. For most animals, bilateral ablations of the primary auditory area produced amnesia, though occasionally this did not occur. (Presumably, the lack of even temporary amnesia reflects incomplete ablations, though final histopathological evaluations are not yet available.) For some animals, amnesia proved to be temporary in the sense that the discrimination tasks could be relearned (though they did not reappear spontaneously), while for others, discrimination performance was permanently affected. The various tasks can be ranked in terms of the probability that the lesion would produce either temporary or permanent amnesia, and these ranks are related to the initial ease of learning. However, since significant individual differences appeared in the ease of initial learning the various discrimination tasks, such a ranking is only of limited accuracy for the group of experimental animals as a whole. It was also found that varying the manner in which the discrimination tasks were structured often significantly altered the accuracy of postablation performance. It is clear that some type of explanative principle similar to Neff's theory of neural habituation for ablated animals is needed.

W11. Duration and Rise Time of Tone Bursts and the Human Vertex Potentials. S. ONISHI (nonmember) AND H. DAVIS, *Central Institute for the Deaf, St. Louis, Missouri.*—Tone bursts with linear rise times of 3 or 30 msec and plateaus from 0 to 300 msec were presented at four intensities to five adult subjects. With a 30-msec rise, the amplitude (N_1 to P_1 and latency (N_1 peak) were unaffected by increasing the plateau from 0 to 300 msec. With a 3-msec rise and 30-msec plateau, the amplitudes were slightly increased at all intensities over the 30-msec rise time. With a 3-msec rise and plateaus of 10, 3, and 0 msec (a tone pip), amplitudes were significantly reduced. The latency was slightly prolonged (*re* 65 and 85 dB) at 45 dB (ISO), significantly prolonged at 25 dB. In another experiment, rise and fall times were varied with a 2.5-sec plateau. Only small differences in amplitude and latency appeared for rise times of 50 msec or less. The increase of latency was one-sixth of the rise time for plateaus at 85 and 65 dB and one-third of the rise time at 45 dB. The same relations hold for the OFF responses although amplitudes were smaller one-third and latencies shorter than for corresponding ON responses. [Work supported by National Institutes of Health, U. S. Department of Health, Education, and Welfare.]

W12. Amplitude of Auditory Evoked Vertex Potentials as a Function of Interstimulus Interval During Slow Repetitive Stimulation. DAVID A. NELSON AND FRANK M. LASSMAN, *University of Minnesota, Minneapolis, Minnesota.*—Averaged auditory evoked potentials were recorded from human vertex in response to repetitive tone bursts presented at different interstimulus intervals. Interstimulus intervals were varied in discrete steps from 0.25 to 10 sec with a counterbalanced order between subjects. Tone bursts were 20-msec pulsed pure tones with 20-msec rise and decay times. Frequencies used were 500, 1000, and 2000 Hz. Although large variance exists in averaged evoked response amplitudes (N_1-P_2 peaks) within and between subjects, group means show N_1-P_2 amplitude to be an increasing function of interstimulus interval: as interstimulus intervals are changed from 0.25 sec to about 2.0 sec, response amplitude increases sharply to a partial asymptote between interstimulus intervals of 2.0 and 3.0 sec, then from about 3.0-sec through at least 10-sec interstimulus intervals, response amplitude continues to increase but the slope of the function is greatly reduced.

THURSDAY, 16 NOVEMBER 1967 EMPIRE ROOM, 9:00 A.M.

Session X. Shock and Vibration I: Geoacoustics

C. W. HORTON, *Chairman*

Invited Papers (25 minutes)

X1. Motion Sensing with Liquid-Filled Systems. DALE W. EVERTSON, *Defense Research Laboratory, The University of Texas at Austin, Austin, Texas.*—Several variables influence hydroacoustic element properties. The dynamics of a motion-sensing system using such elements can be modified more easily than that of a system using mechanical elements, whereas transduction is less difficult using mechanical elements. Examples of hydroacoustics systems for measurement of earthquake motion will be described. Experiences with a hydroacoustic system using a polarized-cathode solion transducer will be discussed.

X2. Short-Period Ocean-Bottom Seismograph—Development and Application. RALPH R. GUIDROZ (nonmember), *Earth Sciences Department, Texas Instruments, Inc., Dallas, Texas.*—Natural and man-made seismic activity in ocean environments can be recorded *in situ* by self-contained ocean-bottom seismograph stations. Development of such operational capability was accomplished during 5 yr of testing and evaluation of three generations of ocean bottom systems. The first and second generation systems were attached by cable to a surface buoy and had less than 12-h recording capability. The present system is allowed to free-fall to operational depths of 4100 fathoms without the requirement of cable or attachment to the surface. The unit records three components of seismic activity and water pressure in the band 1–10 Hz on a 30-day tape recorder. Signals are amplified by low-frequency parametric amplifiers. A digital clock with a timing accuracy of 0.5 sec within a 30-day period is used for timing of data. Release of the system is accomplished by burning a fuse wire. Upon release, the sphere rises to the surface, where a beacon light and radio transmitter are activated to assist recovery operations. With existing technology other parameters can be measured; such as, salinity, temperature, ambient light, sound velocity, depth, gravity, magnetics, and water currents.

X3. Geoacoustic Applications of Electro-Seismic Sources. DAVID M. NASH, JR. (nonmember), AND THOMAS G. BARNES, *Advanced Projects Division, Globe Exploration Co., Inc., El Paso, Texas.*—A mobile high-energy source for electrically producing and controlling seismic pulses is described. Applications that employ one or more of these electroseismic sources include: (1) profiling of stratigraphy anomalies by complete insonification of the target, and (2) propagation investigations associated with a seismic network for detecting missile reentry. The high-frequency characteristics present in the source and the controllability of those characteristics is cited as a means, when used with appropriate sensors, of expanding the application to problems requiring greater resolution of detail.

X4. Rôle of Digital Computers in Exploration Seismology. ROGER D. JUDSON (nonmember), *Chevron Oil Co., Geophysical Division, Houston, Texas* AND J. W. C. SHERWOOD, *Chevron Research Company, La Habra Laboratory, La Habra, California.*—In exploration, seismology surface sound sources and detectors are utilized to indicate variations in the elastic properties of the subsurface of the earth and from these infer the geological structure. It is helpful to pose the basic problem in a form that is common to most scientific investigations. One possesses data (time series from known seismic sources and detectors) plus a conceptual model (elastic wave propagation) with unknown parameters (the elastic variables as a function of space). The problem is to estimate the unknown parameters so as to get some best fit between the model and the data. At the present time, it seems impractical to attempt to solve this total problem directly. Instead, conceptual approximations are made that lead to a number of consecutive problems, each of which normally involves input data, some particular simple model and unknown parameters that need to be estimated before providing output data. A digital computer is a flexible device with which it is feasible to fit a variety of simple models to data, estimate parameters and transform input data to output data. This has caused a major revolution in the processing of seismic data during the last few years and has undoubtedly improved the over-all effectiveness of seismic exploration. Examples will be given of some of the processing.

X5. Seismic Space-Time Processing. MILO M. BACKUS, *Science Services Division, Texas Instruments Inc., Dallas, Texas.*—Frequency filtering and spatial filtering have been usefully applied for many years to the problem of extraction of signals from noise in active seismic exploration and in passive earthquake surveillance. Because broadband (octave or more) transients are of interest in both areas, frequency dependent spatial filtering has been frequently used. In some cases, interfering noise is spatially nonrandom, and "optimum" space-time filters are used to advantage. Essentially similar spatial response is usually obtained independent of the "optimization" criteria (e.g., least squares, maximum likelihood). Similar techniques may be used to employ arrays with complex inter-element signal relationships (such as combinations of horizontal and vertical motion seismometers). Although useful coherent properties of signal and noise can be exploited with an assumption of space stationarity, time stationarity, or space-time stationarity, in many cases use of space-time varying optimization is worthwhile and is being increasingly employed. Some of the current approaches to tracking down useful statistical space-time properties of signal and noise and their exploitation in seismic information extraction will be discussed.

Contributed Papers (12 minutes)

X6. Source Size as a Theoretical Limitation on the Determination of Wave Vectors by Detector Arrays. ERIC POSMENTIER, *Lamont Geological Observatory, Columbia University, Palisades, New York 10964.*—An important class of infrasonic waves in the atmosphere and seismic waves in the earth's crust are generated by ocean waves in marine storms. The determination of wave vectors from such large sources by detector arrays, is limited by the decrease in resolution, ambiguities, and systematic errors introduced by the large sources, as well as by the resolving power and ambiguities associated with a given array configuration. The apparent phase lag and coherency between two detectors, each receiving signals from the multitude of mutually incoherent oscillators comprising a large source, is discussed as a function of source size, and source-detector configuration. Several examples of source-array configurations are discussed, and calculated values of the discrepancies between apparent wave vectors and wave vectors from the source centers are presented. Empirical results are presented to confirm the validity of this approach in the case of microbarom generation by ocean waves.

THURSDAY, 16 NOVEMBER 1967 — MEDALLION WEST, 9:00 A.M.

Session Y. Speech III: Hearing and Transmission

PETER B. DENES, *Chairman*

Contributed Papers (10 minutes)

Y1. Some Temporal Relationships between Speech and Hearing. JIM D. MCDILL (nonmember) AND ROBERT B. MAHAFFEY, *The University of Southern Mississippi, Mississippi.*—This study was designed to explore whether certain temporal patterns of speech are related to temporal aspects of hearing. The results of the study were obtained by correlating the tone-burst durations most accurately matched at each of three frequencies and the critical fusion frequency with three measures of speech. These measures were average phonation time, average pause time and the number of pauses per second. Twenty-one male and 21 female persons with normal hearing served as subjects. The results indicate that significant relationships exist between the average pause time and critical fusion frequency and tone-burst duration most accurately matched at 150 Hz. The same relationships existed for the average phonation time. These relationships would indicate a possible dependency of speech timing upon certain primitive aspects of temporal processing by the auditory system.

Y2. Discrimination of Vowel Spectrum in Sensorineural Hearing Impairment. J. M. PICKETT AND ELLEN S. MARTIN, *Hearing and Speech Center, Gallaudet College, Washington, D. C. 20002.*—Measurements are reported comparing normal and impaired auditory discrimination for formant frequencies of synthetic vowels. Vowel spectra were shaped by two formant resonators with controllable center frequencies, $F1$ and $F2$. For discrimination measurements three vowels were presented on each trial. For two of the vowels, $F1$ and $F2$ were held constant. For the third vowel, $F2$ was set at a higher frequency. The task of the listener was to tell which one of the vowels was different from the other two. A programmer controlled the synthesizer so as to present a sequence of trials with downward adjustment of amount of $F2$ difference after each correct response and upward adjustment after each wrong response. Thus, over a sequence of trials the amount of $F2$ difference typically decreased from above the difference limen until oscillatory adjustments occurred above and below the limen. Results: frequency position and relative amplitude of $F1$

has a marked effect of the $F2$ discrimination limen. All groups show good $F2$ discrimination when the formants are separated by more than 1 oct. Some subjects demonstrated abnormally poor discrimination when $F2$ was 10 dB lower than $F1$, but improved when the amplitude difference was reduced. [Work supported by U. S. Public Health Service grant.]

Y3. Experiments on the Use of the Touch-Tone Telephone as a Communication Aid for the Deaf. J. R. NELSON (nonmember), *Bell Telephone Laboratories, Incorporated, Murray Hill, New Jersey.*—Two methods are described for using the Touch-Tone telephone as a communication aid for the deaf. Both methods depend upon the calling party spelling the message to the deaf receiver using a previously agreed upon code. Both methods also require circuit attachments to the conventional Touch-Tone telephone set of the receiver. In the first method, an auxiliary Touch-Tone dial is available at the receiver. Depressing a Touch-Tone button at the transmitter causes the corresponding button to light on the receiver's auxiliary dial. The auxiliary circuitry for visual presentation at the receiver includes standard Touch-Tone decoding filters and simple AND gates. The second method utilizes the same transmission technique but has more sophisticated circuitry at the receiver to accomplish the multiple-character decoding automatically. In this case, the visual presentation is the alphanumeric character which was transmitted. Experiments are described on the speed and accuracy with which these techniques can be used. Typical communication speeds are 4 words/min for the first method and 8 words/min for the second method. Both methods yield comparable error percentages, on the order of 0.2%. This performance is achieved without special training or skill.

Y4. Index of Criterion for Sentence Identification Tasks. CHARLES SPEAKS, *Resea ch Institute of the Houston Speech and Hearing Center, Houston, Texas 77025.*—At a previous meeting, the applicability of a statistical-decision model to a sentence identification task was described. The observer's task is to decide if a message belongs to a primary or secondary subset of messages. If primary, he is to decide which alternative primary message was presented. The observers generated ROC curves consistent with the model. A fortunate by product of the model is that in addition to describing performance by a criterion-free measure, it also is possible to specify, quantitatively, the criterion adopted by the observer for the task. The criterial index yields a value on a linear scale that ranges from -0.50 (extreme strict criterion) to $+0.50$ (extreme lax criterion). The theoretical distributions of the index for various values of d' and empirical data from hearing-impaired persons will be presented.

Y5. Effects of Delayed Sidetone on the Speech of Aphasic, Dysarthric, and Mentally Retarded Subjects. SADANAND SINGH AND BERNARD B. SCHLANGER (nonmember), *The Ohio State University, Columbus, Ohio.*—Duration, amplitude, and phonetic errors were measured to test the effects of delayed and normal sidetones on the speech of 10 aphasic, 10 dysarthric, and 10 mentally retarded subjects. Additional variables included structurally correct semantically real and unreal kernel, negative, query, and negative-query sentences. Tested were the differences between the three groups of subjects under conditions of 0.18-sec delay and normal delay in sidetone, three degrees of grammaticalness, and the four syntactic classes. The delayed sidetone affected the mentally retarded and dysarthric subjects more than the aphasics. The ratio between the duration under the 0.18-sec delayed sidetone and normal sidetone conditions for aphasic subjects was 1.20:1, for dysarthric 1.62:1, and for mentally retarded, 1.72:1. The grammatical sentences were affected less than the ones with unreal verbs. Sentences with unreal nouns, verbs, and objects were maximally affected. The grammatical classes were affected from least to the most in this order: kernel, query, negative, and negative-query.

Y6. Evaluation and Optimization of Nonlinear Quantization Characteristics in PCM Transmission of Speech. HIROYA FUJISAKI AND MASANORI KATAOKA (nonmember), *Engineering Research Institute, Faculty of Enginee ing, University of Tokyo, Bunkyo-ku, Tokyo, Japan.*—Nonlinear quantization has been introduced into PCM transmission of speech in order to reduce the quantization effects. Although signal-to-noise ratio has been conventionally adopted for the evaluation of such systems, assuming a negative exponential distribution for speech amplitude, there is little evidence for its validity. An evaluation theory is presented based on the information transmission rate of a quantized channel and approximating the temporal structure of speech by two distributions corresponding to voiced and unvoiced sounds. Articulation tests have been performed using both actual PCM systems and a flexible simulator and the results have been analyzed both in terms of the articulation score and the amount of information transmitted per syllable. The results indicate poor correlation between conventional calculations and articulation scores, and prove the validity of the proposed theory. Comparisons of known nonlinear characteristics lead to an optimum one that is somewhere between the logarithmic and the hyperbolic characteristics.

Y7. Distribution Theory in Vocal Tract Analysis. JERRY A. SELVAGGI (nonmember), *Gannon College, Erie, Pennslyvania* AND JOSEPH L. DE CLERK (nonmember), *US Army Electronics Command, Fort Monmouth, New Jersey.*—Solution of Webster's horn equation is accomplished in a discrete manner by means of distribution on theory. In the process of applying distribution theory exact solutions result. In the past, numerical integration was utilized. Given output formant frequencies, area functions can be calculated, or conversely, formant frequencies can be calculated from vocal tract area data. The results can be converted to an analog, which describes the behavior of the vocal tract. By quantizing the vocal tract an exact solution to Webster's horn equation results, whose reciprocals turn out to be transfer functions of transmission line theory under certain boundary conditions. Radiation impedance has been absorbed in the quantized vocal tract sections. Actual area functions of the vocal tract are determined by midsagittal cinefluorographic tracings and dental casts of the mouth. Acoustic data are synchronized with the cinefluorographic data by means of a digital code that provides registration between articulatory events and formant tracks. Determining vocal tract analogs by physiological data is more desirable than using formant information alone since consonants as well as vowels can be included in the analog. In addition, by referring to data that change every 8.33 msec a better insight is gained concerning articulatory movement. This technique is more realistic since it uses dynamic data which more nearly typifies the transient characteristics of speech than the usual steady-state analysis.

Y8. Differential Effects on Underwater Hearing Thresholds of Air and Water Interface with the Tympanic Membrane. HARRY HOLLIEN, *Communication Sciences Laboratory, University of Florida, Gainesville, Florida.*—Thresholds of human hearing were obtained underwater for two conditions: (1) with the external auditory meatus completely water filled and (2) with a bubble of air trapped against the tympanic membrane. The first condition was accomplished by an otolaryngologist forcibly irrigating the external meatus *under water*; the second by placing plugs in the ears in order to encapsulate air in the meatus until the head was underwater and the test (with the plugs removed) initiated. Subjects were seven divers who were tested in DICORS at a depth of

12 ft. Threshold SPL's were obtained freefield by the Békésy technique for the frequencies 125, 250, 1K, 2K, and 8Hz. Threshold shifts (*re:* air) for both conditions of underwater hearing were consistant with those previously reported. SPL's for the two experimental conditions were virtually identical for all frequencies except 250 Hz where hearing was 6dB better for the water-filled meatus condition. Statistical analysis (AV) of the data showed lack of significance at the 5% level of confidence. [The project was supported by the Office of Naval Research.]

Y9. Study of Relations Among Terms Used for Subjective Evaluation of Telephone Transmission Quality. J. P. DUNCANSON (nonmember), *Bell Telephone Laboratories, Inc.*—Many different rating scales have been used in the study of telephone transmission quality (for example: excellent, good, fair, poor). Twenty-one terms from five such scales were studied using two experimental techniques, paired comparisons and sorting by similarity of meaning. The first method yielded a scale value for each term along a single dimension of transmission quality. These values revealed that the five original rating scales vary widely in the range of quality spanned and in the evenness of spacing of the steps within the scale. The sorting method yielded a hierarchial clustering of the terms. It was found that the 21 terms can be sorted into five groups, the terms within a group being more or less equivalent in meaning. These two sets of data can be useful in constructing rating scales for particular applications by permitting (1) selection of terms to cover the desired range of quality, (2) selection of terms spaced approximately equally over the desired range, and (3) construction of two or more equivalent forms for use in repeated or reliability testing.

Y10. Objective Procedure for Reading VU Meters. HARRY LEVITT AND PETER D. BRICKER, *Bell Telephone Laboratories, Incorporated, Murray Hill, New Jersey 07971.*—Speech level measurements obtained with a VU meter are subject to several sources of variability. The measuring procedure outlined here is designed to reduce variability introduced by the meter reader and so provide a reading more representative of average speech level. A sampling procedure is used in which controlled illumination renders the meter pointer visible only during brief intervals. Each time the dial is illuminated, the meter reader is required to determine whether or not the pointer exceeds a given deflection. If the pointer exceeds the critical value, the speech signal is attenuated by a fixed amount before the next trial. If the pointer lies below the critical value, the attenuation is reduced for the next trial. An up-down procedure with a regularly decreasing step size is used. The technique is such that the attenuation level converges rapidly on that value at which the pointer exceeds the critical value a fixed proportion of the time. Measurements obtained with this procedure have been found to be highly repeatable and relatively free of intersubject differences.

Y11. Vocoder Filter Design: Practical Considerations. ROGER M. GOLDEN, *Autonetics, Division of North American Aviation, Anaheim, Calif. 92803.*—Successful digital computer simulation of a spectrum channel vocoder, [R. M. Golden, J. Acoust. Soc. Am. **35**, 805(A) (1963)] has also led to new filter design procedures for vocoder bandpass filters. These procedures yield designs that result in improved performance of the spectrum channel vocoder. The procedures are accomplished by computer aided design programs. The programs permit the examination and modification of the time and frequency response characteristics of the various bandpass and low-pass filters required by a spectrum channel vocoder. Such examination readily shows why the Bessel (or Linear Phase) designs are to be preferred over filter designs such as Butterworth and Elliptic. In addition to choice of prototype filter design, consideration must also be given to filter resolving power (i.e., passband and stop-band characteristics) and to the frequency cross-over point of contiguous filters. This latter consideration accounts for the fact that performance of some vocoders has been improved by the simple device of shifting the phase of alternate channels by 180°. For many vocoder configurations, equalization of filter group delay characteristics is necessary between various channels. This factor is important for those vocoder designs using increasing bandwidth filters for the higher frequency channels. Equal bandwidth filters combined in pairs or triplets achieve both equalization and the desired increase in bandwidth.

Y12. New Artificial Voice and Applications. RICHARD W. CARLISLE, *Dyna Magnetic Devices, Inc., Hicksville, New York, 11801.*—The artificial voice to be described is an elaboration of the WE 555 Speaker and coupler utilized in current Standards. A felt damper pad and an acoustic tube load, with damping in the tube, have been added. These lower the resonant frequency and smooth midrange and high frequencies. The mouth is oblate in shape. The distance from the felt damper, and the effective area of the sound passage to the mouth, are estimated to be of the same order of magnitude as corresponding portions of the human vocal tract. Application of this artificial voice to sound radiation from a simply proportioned plastic face are described, together with application of the ensemble to measurements within an oxygen mask.

THURSDAY, 16 NOVEMBER 1967 — MEDALLION EAST, 2:00 P.M.

Session Z. Psychological and Physiological Acoustics VI: Masking and Detection

DONALD ROBINSON, *Chairman*

Contributed Papers (10 minutes)

Z1. Ohm's Law and Masking. B. L. CARDOZO, *Institute for Perception Research, Eindhoven, Netherlands.*—The harmonics in the spectrum of a periodic pulse train should, according to Ohm's law, be audible separately. When trying to check this one meets two difficulties in terms of subjective spectrum analysis. First, the *n*th harmonic may be masked in part or completely by other harmonics. Second, even if the *n*th harmonic is not masked completely, one has to *find* it among a great many others. Makins use of "Schouten's artifice" (alternatively suppressing and reintroducing the *n*th harmonic), these difficulties can be lessened. Once the *n*th harmonic is heard, one may determine its masked threshold by attenuating it. Using the method of adjustment, three experienced subjects measured the masked thresholds of the lowest 10–25 harmonics of a 50, 100, 200, and 400 Hz periodic pulse train at 30 dB SL. From these measurements masking curves are derived. The slopes of these curves show reasonable agreement with those of earlier investigations. A feature of the present method is that one need not make use of noise and still is able to determine masking of closely adjacent frequencies. In varying the time interval, the experment also shed some light on the dynamics of masking.

Z2. Detection of Tones in the Absence of External Masking Noise. CHARLES S. WATSON, JOHN R. FRANKS (nonmember), AND DONALD C. HOOD (nonmember), *Central Institute for the Deaf, St. Louis, Missouri*.—Sound pressures required for various levels of detection performance, in the absence of external masking, are summarized in iso-detectability contours. Each such contour shows, for frequencies from 125 to 4000 Hz, the SPL of 150-msec tone pulses which yield a specific probability of correct decisions in a two-alternative, temporal, forced-choice psychophysical procedure (2ATFC). Detection in the absence of externally introduced masking noise is found to differ in no qualitative ways from that in the presence of noise. Additional results suggest a spectrum of "internal noise" which is essentially a critical ratio below the sound pressure level of low-detectability ($d'=1.0$) tones in the quiet. Iso-detectability contours for detection performance of 55% and 99% correct in the 2ATFC procedure are found to enclose the range of sound pressure levels currently accepted as audiometric zero (ISO). [Work supported by the National Institutes of Health, Public Health Service, U. S. Department of Health, Education and Welfare.]

Z3. Shifts in Masking with Time. ROBERT C. BILGER, *Bioacoustics Laboratory, University of Pittsburgh, Pittsburgh, Pennsylvania*, AND WILLIAM MELNICK, *Ohio State University, Columbus, Ohio*.—A time shift of remotely masked thresholds during the first 2 min after onset of noise has been contrasted to a presumed invariance of directly masked thresholds over time by earlier investigators. The time course of masking of a 500-Hz signal was determined for remote masking (noise from 2000–4000 Hz) and for direct masking (noise from 200–4000 Hz). Masking was determined for simultaneous onset of the noise and test signal and for delayed (2 min) onset of the tracking task. For half of the conditions, listeners varied level of the tone against a constant background noise, and in the other half they varied noise level to mask a constant level of pure tone. The results, in terms of signal-to-noise ratios, show that a downward shift in directly masked threshold (3 dB) occurred during the first 2 min of tracking. This shift in direct masking was influenced by both time of onset of the tracking task and by mode of control. The shift in remote masking with time (4.5 dB) in 2 min was independent of both onset and control.

Z4. Better than Energy Detector Performance by Human Observers. M. M. TAYLOR AND S. M. FORBES (nonmember), *Defense Research Establishment Toronto, P. O. Box 2000, Downsview, Ontario*.—Observers detected, monaurally, a burst of band-limited noise masked by an independently generated, continuously presented, wide-band noise. An unmasked burst of noise in the other ear served as a cue. Four cue conditions and several signal bandwidths were used. The cue noise burst had the same nominal spectrum, amplitude, and timing as the signal burst, and occurred in each of the two intervals of a 2AFC trial. The four cue conditions were: no cue (O); independently generated cue burst (I); cue identical to the signal (S); cue derived from the signal passed through a wideband 90° phase shifter (X). Potentially, I provides timing information, X supplies information about short-term fluctuations of signal spectrum and amplitude, and S gives complete information about the signal waveform. At all signal bandwidths, the use of O and I resulted in performance levels slightly worse than an energy detector; X allowed much better performance. Results with S approached that of the $(2E/N_0)^{\frac{1}{2}}$ detector, which at large signal bandwidths is as much as 18 dB better than an energy detector.

Z5. Agreement in Detection: Observers and Electrical Model. PAUL I. WILLIAMS AND LLOYD A. JEFFRESS, *Department of Psychology and Defense Research Laboratory, The University of Texas, Austin, Texas 78712*.—At the Los Angeles meeting, Nichols reported that an electrical model (narrow filter-rectifier-envelope filter) predicted the responses of four observers in a single-interval detection experiment better than either a different electrical device (envelope-maximum detector) or the presence or absence of the signal. He suggested the desirability of employing multiple signal levels and a two-alternative, forced-choice method with a similar detector. The present paper reports such an experiment, with the first electrical model and three observers responding to the same stimuli. Seven signal levels were employed, and on some trials, noise alone occurred in both intervals. $P(C)$ ranged from about 0.99 to chance (for noise-alone trials). The data show that the model predicts observers' behavior better than the presence or absence of the signal does, and better than one observer predicts another. When the three observers agreed in their responses (whether right or wrong) the model agreed with them on more than 81% of the trials. The weaker the signal the greater was the superiority of the model over the signal as predictor. (Noise: 100–3000 Hz, 50-dB spectral level. Signal: 500 Hz, 0.25 sec.) [Supported by the U. S. Naval ship Systems Command and the National Aeronautics and Space Administration.]

Z6. Signal Detection with a Fixed Sample of Masking Noise. JOSEPH MARKOWITZ AND JOHN A. SWETS, *Bolt Beranek and Newman Inc., Cambridge, Massachusetts 02138*.—Detection of sinusoidal signals over trials containing precisely the same sample of background noise ("same-noise" condition) was compared with detection over trials in which the sample of noise varied from one trial to another ("different-noise" condition). All noise samples had a duration of 0.1 sec and a bandwidth of approximately 5000 cps. Differences among individual observers, with respect to the relative levels of detectability produced by the two conditions, were marked, and were consistent across several signal-to-noise levels, and across three psychophysical procedures. Preliminary results suggest that the individual differences in detection performance are correlated with differences in the individuals' abilities, assessed in separate tests, to discriminate among the various noise samples used. An observer able rather easily to distinguish different noise samples was substantially better at detecting the sinusoidal signal in the same-noise condition than in the different-noise condition; an observer almost completely unable to discriminate among various noise samples was poorer at detecting sinusoids in the same-noise condition than in the different-noise condition. [This work was supported by the Human Performance Branch of the NASA Ames Research Center.]

Z7. Signal Uncertainty in the Multiple-Observation Detection Task. JOHN A. SWETS AND JOSEPH MARKOWITZ, *Bolt Beranek and Newman Inc., Cambridge, Massachusetts 02138*.—Rates of growth of signal detectability, throughout trials containing several successive observation intervals, were compared for two conditions of uncertainty about signal frequency. The signal was a burst of either 700 or 1700 cps for 0.1 sec. The signal frequency was randomly varied *within* a trial ("simple-hypothesis" condition) or only *between* trials ("composite-hypothesis" condition). If, in the composite condition, the observer determined early in the trial which frequency was being presented, and used this knowledge to advantage in the later observation intervals of the trial, we should have obtained a rate of growth in detectability in the composite condition that was greater than the rate obtained in the simple condition, where no reduction of uncertainty was possible. In fact, there was no discernible difference in detectability in the two conditions, suggesting that the observers did not observe and accumulate information separately at the two frequencies. Nonetheless, the observer's ability to recognize which of the two frequencies was present in the composite condition increased substantially over the first few observation intervals of a trial. [This work was supported by the Human Performance Branch of the NASA Ames Research Center.]

Z8. Effects of Signal Probability on the Listening Band. ROBERT D. SORKIN, *Purdue University, Lafayette, Indiana*—A two-interval detection procedure was used to study the effects of signal probability on the listening band. The signal was chosen randomly on each trial from a set of three 100-msec, equal amplitude, phase-coherent sinusoid segments of 650, 750, or 850 Hz. The relative likelihood of a specific frequency signal being present on a trail was varied over three experimental conditions: (1) $P(650)=P(750)=P(850)=0.33$; (2) $P(650)=P(850)=0.17$, $P(750)=0.66$; and (3) $P(750)=1.0$. Performance was assessed under constant noise level and two signal levels ($10 \log E/No=10.4$ and 12.2 dB). Plots of $P(C)$-vs-signal frequency indicate a small but consistent difference between the three experimental conditions. The effective listening band appears to be sharper under conditions in which the center frequency signal has a higher presentation probability. Analysis of the one trial sequential probabilities fails to support an interpretation of the results in terms of a simple sequential model. The observers appear to be using correct-trial information in order to improve their ability to detect the midfrequency signal, even when the preceding correct-trial was on a high (or low) signal.

Z9. Sequential Dependence in Signal Detection, and Parameter Estimation by Sequential Tracking (PEST). M. M. TAYLOR, S. M. FORBES (nonmember), AND C. D. CREELMAN, *Defence Research Establishment Toronto, P. O. Box 2000, Downsview, Ontario*.—The PEST psychophysical method [Taylor and Creelman, J. Acoust. Soc. Am. (to be published)] was used to determine the signal burst intensity giving 80% correct responses in a 2AFC detection task. Each PEST run was immediately followed by fixed level trials at that level. For the fixed level trials, $P(C)$ averaged only 0.75. However, the probability of a correct response conditional on a correct preceding response, $P(C/C_{-1})$, averaged 0.765. Assuming the apparent correlation to result from averaging results over periods of good and poor detection, the worst possible performance during "good" periods is about 0.80. These results suggest that PEST may be more accurate than are the conventional fixed level methods. PEST makes the task easier when the subject "loses track" of what he is trying to detect; whereas in a fixed level experiment the subject has little opportunity to refresh his memory for the signal to be detected. In computer simulation, PEST has proved to be quick and unbiased.

Z10. Measures of Performance Sensitivity. GORDON W. WILCOX, *Department of Psychology, Cornell University, Ithaca, New York*.—Several writers have suggested fitting straight lines on double-probability paper to the empirically obtained receiver-operating-characteristic (ROC) curves for human observers. A simple theorem is proved which shows that in acoustical tests in which a fixed signal is presented, *no* observer of any kind can have a straight-line ROC with other than unity slope. This negative result suggests that some commonly used measures of performance sensitivity are inadequate. A new interpretation of Green's theorem relating the area under the single-interval ROC to the probability of correct response in two-interval forced-choice tasks is given. The interpretation permits a new and direct estimate of the area under the ROC curve in single-interval rating tasks. The estimator, denoted P_A, (i) is unbiased, (ii) depends only upon obtained ROC points (i.e., no curve must be fitted to the data), and (iii) generalizes to certain multiple signal-alternative situations. It is further shown that P_A is a special case of a more general measure of association between the performance of any two observers (human or electronic) in the same task. As such, it may be used in testing alternative models of sensory processing [This reserach was conducted at the Sensory Intelligence Laboratory, University of Michigan, and supported by a contract with NASA.]

Z11. Statistical Estimation of Parameters of Decision-Theory Models. INNIS ABRAHAMSON (nonmember), HARRY LEVITT, AND LORINDA LANDGRAF (nonmember), *Bell Telephone Laboratories, Incorporated, Murray Hill, New Jersey 07971*.—Statistical procedures have been developed for estimating the parameters of decision-theory models. The problem reduces essentially to that of fitting receiver-operating-characteristic (ROC) curves to experimental data. Difficulties arise in that both coordinates of each plotted point are subject to errors of estimation. It is assumed that the likelihood distributions for noise alone and for signal plus noise belong to the same location-scale family (e.g., both normal) with standard deviations σ and σ_S, respectively. It is further assumed that the samples are sufficiently large to warrant a normal approximation to the multinomial distribution. The analysis has been carried out for YES–NO, forced-choice and rating-scale data, both for $\sigma=\sigma_S$ and $\sigma\neq\sigma_S$. Bias and precision of estimation have been considered as well as procedures for testing the underlying assumptions of the model. A Monte Carlo study has been carried out to investigate the quality of large sample procedures based on the method of maximum likelihood and to compare them with several simplified approximate procedures.

Z12. TSD Applied to Footfalls in Buildings. B. G. WATTERS, *Bolt Beranek and Newman Inc*.—We have studied the acceptability of intruding footfalls in buildings, and found that "just-detectable" signals cause significant annoyance. Consequently, we have considered how well the theory of signal detectability (TDS) predicts the perception of footfall sounds modified by low-pass filtering and reverberation, and partially masked by random noise. The "heel click" portion of a footfall signature, often the acoustically important part, has been simulated by a short series of periodic, filtered Dirac impulses. The signal replication caused by multiple echoes in rooms was progressively introduced. The first complication was a series of three of the above impulses, spaced by 10 msec and heard anechoically. The next was a reverberant signal, generated by producing single clicks in a $\frac{1}{2}$-sec RT room, picking these up with a microphone and reproducing them as unidirectional reverberant pulse trains in the anechoic chamber. Finally, the subjects were placed directly in the reverberant room. All the above signals are recognizable as (synthetic) footfalls. We will discuss whether the memory which permits the subject to recognize these waveforms is adequate to enable him to perform cross correlation in the detection process, as for the SKE and SKEP models. [This work was supported by the Armstrong Cork Company.]

THURSDAY, 16 NOVEMBER 1967 BURGUNDY EAST, 2:00 P.M.

Session AA. Engineering Acoustics III: Radiation

MIGUEL JUNGER, *Chairman*

Contributed Papers (10 minutes)

AA1. Acoustic Field Due to a Partially Excited Spherical Radiator. K. L. MILLER (nonmember) AND C. T. MOLLOY, *TRW Systems Group, Redondo Beach, California.*—In previous papers, it was shown how to obtain prescribed acoustic pressure fields by properly choosing the radial velocity distribution on the surface of a spherical transducer. In all of these cases, the transducer was required to radiate from its entire surface. In the present paper the effects of allowing the transducer to be activated only over a portion of its surface is investigated. A desired directivity pattern of the Chebyshev type is selected. The transducer velocity distribution that will produce it is then calculated. This distribution, which is symmetric about the transducer axis, and covers the whole sphere ($0\leq\theta\leq\pi$) is then successively truncated so that a sequence of velocity distributions is produced each of which is identical with the original distribution up to some angle (θ_n) and is zero for $\theta>\theta_n$. The directivity patterns associated with each of these truncated distributions was calculated. The results to date indicate that the Chebyshev-type distribution permits substantial velocity distribution truncation without serious degradation of the directivity pattern. Several numerical examples will be presented. Some results obtained by truncating distributions of the $\cos^n\theta$ type will also be presented.

AA2. Nonuniform Spherical Radiation Through Thick Shells. G. C. K. YEH (nonmember) AND C. T. MOLLOY, *TRW Systems Group, Redondo Beach, California 90278.*—The work previously done on uniform radiation through thick spherical shells has been extended to cover arbitrary, nonuniform radiation. A spherical transducer is surrounded by a fluid medium that is in turn surrounded by a thick viscoelastic shell. The shell is immersed in a fluid medium of infinite extent. Functions have been derived that define the pressure field due to the transducer that is driven sinusoidally with a prescribed radial velocity amplitude over its surface. The inverse problem is also solved, namely: to find the radial velocity distribution on the transducer which will produce a prescribed pressure field. These results have been used to study the effect of the shell upon the directivity pattern of the transducer and the effect of the shell upon the transducer velocity pattern when it is producing a prescribed pressure field. The insertion loss due to the shell has also been examined. Numerical calculations illustrating these effects are presented.

AA3. Radiation of Pistons and Rings on Oblate and Prolate Spheroidal Baffles. MARVIN A. BLIZARD (nonmember) AND JAMES F. DILLON (nonmember), *Naval Research Laboratory Washington, D. C. 20390.*—The radiation impedance, self and mutual, and nearfield pressure of finite pistons and rings on oblate and prolate spheroidal baffles have been computed for several combinations of radiator size, ratio of height-to-diameter (α), and of the frequency parameter $h(h=Kd/2)$, where K is the wavenumber and d is the interfocal distance of the spheroid). These results were obtained by substituting numerical values of the oblate and prolate spheroidal wavefunctions (recently computed at NRL by Hanish *et al.*) in the infinite series solutions of Nimura and Watanabe. For the oblate case, $0.1\leq h\leq 12.0$ and $0\leq\alpha\leq 0.8944$ (Nimura and Watanabe gave results for $0.01\leq h\leq 4.0$ and $\alpha=0$). For the prolate case, $0.1\leq h\leq 10.0$ and $1.005\leq\alpha\leq 10.0$ (Nimura and Watanabe gave results for $0.06\leq h\leq 6.0$ and $\alpha=1.8$). These calculations provide a compilation of numerical values for impedance and nearfield pressure in ranges of h, α, and radiator size, heretofore never obtained from an exact infinite-series solution.

AA4. Numerical Approach to Pattern Synthesis. D. F. LITTLE (nonmember), *Naval Undersea Warfare Center, San Diego, California 92152.*—In the past, many methods of synthesizing patterns of radiated energy have been advanced for special situations. Modern computing techniques make possible a direct approach applicable to arrays of general geometry and to general modes of excitation. Furthermore, prescribed patterns are not constrained to special mathematical forms. For application of the method, it is necessary to obtain experimentally or analytically a transformation matrix—probably nonsquare—which connects boundary values on the array with farfield pressures. The problem is solved by generating the generalized inverse of the matrix. A description and analysis of the method as well as the following results are included in the discussion: (1) side-lobe reduction of a pattern from a band aperture on a sphere was accomplished without substantial changes in output power or radiation impedances; (2) a pattern with a 90° beamwidth and no side lobes was synthesized from a line array; (3) a pattern with a 15° beamwidth and "back radiation" 20 dB down was computed using two columns of cylinders and taking account of individual element patterns.

AA5. Special Relationships between the Farfield and the Radiation Impedance. CHARLES H. SHERMAN, *Parke Mathematical Laboratories, Inc., Carlisle, Massachusetts.*—Some new relationships between the sound pressure at certain points in the farfield and the radiation impedance for certain acoustic sources have been derived from the Helmholtz integral formula. Under special conditions these relationships reduce to exact analytical solutions of problems that can otherwise only be solved numerically. For example, for a cylinder of arbitrary cross sectional shape with the ends vibrating uniformly and the sides motionless the farfield pressure in the direction of the cylinder axis is simply related to the radiation impedance. Such relationships provide helpful checkpoints for evaluating the success of some of the elaborate numerical methods which are currently being developed for attacking previously unsolved acoustic radiation problems with large digital computers. In certain cases, they can also provide new methods for determining radiation impedance from farfield calculations or possibly from farfield measurements. [Work supported by U. S. Navy Underwater Sound Laboratory.]

AA6. Sound Radiation of Complex Vibrators. EUGEN J. SKUDRZYK, *The Pennsylvania State University.*—The simplest type of vibrators are point and line sources. The sound fields of such vibrators are uncorrelated over space and their energies add, if they are separated by a distance greater than about ¼ wavelength. Vibrators with nodal lines radiate as if their radiation impedance were ρc if the distance between the nodal lines is greater than half a sound wavelength. If their distance is smaller, they generate sound because of the distortion of the natural mode patterns by the driving force and because of the discontinuity of the vibration pattern near the edges of the vibrator. Both types of sound radiation turn out to be predictable in a very simple manner by a kind of Huygen's principle. Because ribs also distort the vibration field, they lead to increased sound radiation. [Work sponsored by the Office of Naval Research, Washington, D. C.]

AA7. Numerical Methods for the Computation of Acoustic Radiation Functions, G. E. MARTIN, *Naval Undersea Warfare Center, San Diego, California 92152.*—During the last decade, the large scale digital computer has become available to almost all scientists. The speed and accuracy of such machines has led some people to use mathematical methods that are not always appropriate. The primary purpose of this paper is to discuss some numerical methods for the improvement of the convergence of series encountered in acoustical theories. This paper is restricted to general comments and a discussion of one special case, namely the mutual mechanical impedance between circular pistons in an infinite plane rigid baffle, which has been represented by Pritchard as a double or single series of Bessel function products. If the separation of the two pistons is large as compared with the diameter of the pistons, the series can be represented to sufficient accuracy by its partial sum of a very few terms. On the other hand, if the separation is only slightly greater than the diameter, a large number of terms must be included in the partial sum. Methods for reducing the numerical computations and/or improving the accuracy of results have been studied. An integral method has been used with excellent success.

AA8. Comparison of Methods of Farfield Prediction: DONALD A. MURPHY, *Hughes Aircraft Company, Fullerton, California.*—All farfield prediction methods utilizing nearfield measurements on a closed surface are describable as applications of a Green's integral formula. For truncated measuring surfaces, however, the corrections that must be made to the weighting functions derived from the Green's integral formula can severely limit their usefulness. For the case of spherical and cylindrical measuring surfaces, the corrections for truncated surfaces do not appear to be necessary; but for truncated planar surfaces they are dominant. The broadband "Green's transfer" functions used in farfield prediction for spherical surfaces have been described in a previous paper [J. Acoust. Soc. Am. **39**, 1222(A) (1966)]. The Green's transfer functions for a cylindrical surface have been calculated and farfield predictions made on the digital computer. They are very similar to those for the spherical surface having the same weighting and phasing, although calculated from an altogether different mathematical expression. This expression will be compared to that used by Baker (Univ. of Texas Defense Res. Lab. Acoust. Rept. 196 (15 March 1962) for single-frequency measurements on a cylindrical surface.

AA9. Radiation Characteristics of a Circular Transducer Surrounded by a Nonconcentric Circular Shell. W. C. MOYER AND K. B. HAMILTON, *TRACOR Inc., Austin, Texas 78701.*—An analytical solution is derived for the sound pressure field produced by a cylindrical transducer surrounded by a nonconcentric, circular, shell-type dome. This model represents an improvement over concentric circular dome models previously reported since transmit beams nonsymmetrically incident on the dome can be analyzed. Numerical results indicate two features, (1) the minor lobe structure of the farfield beam patterns is not symmetric relative to the beam axis and is dependent on the eccentricity of the dome and transducer; and (2) oblique incidence of the beam on the dome does not produce beam steering errors, i.e., the maximum response of the beam pattern agrees with the steered direction.

AA10. Propagation of Acoustic Transients in a Fluid Layer. ALAN B. COPPENS, *Naval Postgraduate School, Monterey, California.*—Approximate solutions of the wave equation have been obtained for some acoustic transients propagated radially outward from a cylindrical source of radius a in a fluid-layer with pressure-release (or rigid) surfaces. The earliest portions of the received signals display amplitude- and phase-modulations dependent on the ratio $\omega_c a/c$, where ω_c is the (angular) cutoff frequency for the excited mode and c is the free-field speed of propagation of sound in the fluid. Relations between this problem and (a) propagation of an acoustic transient from a point source in a fluid layer and (b) propagation of a transient in a duct of uniform cross section are pointed out. [Supported in part by Ship Systems Command and Air Systems Command.]

AA11. Wave Propagation across an Interface between Solid and Fluid Bounded Media. G. K. MILLER, II AND E. L. HIXSON, *The University of Texas, Austin, Texas 78712.*—Stress-wave propagation conditions across a plane interface between a cylindrical solid rod and confined fluid column are determined. This is done by expressing the displacement and stress fields as infinite sums over the eigenvalues found from dispersion relations resulting from the appropriate wave equations and transverse boundary conditions. To obtain numerical results, these sums are truncated and the interface boundary conditions are imposed at enough points along a radius to solve for amplitude coefficients in the truncated series. The number of terms can be increased to satisfy the boundary conditions everywhere to an arbitrary degree of accuracy. For the case of nylon to water, fields are calculated in both media from which intensity, power flow, and transmission loss are determined. New information is obtained about the stress and displacement fields as a function of frequency, axial and radial position. An end resonance similar to that of a free bar is found at which practically no energy couples to the water. [Research supported in part by the Office of Naval Research, The Link Foundation, and Department of Defense's Joint Services Electronics Program.]

AA12. Finite Amplitude Distortion of 150-kHz Acoustic Waves in Water. DAVID G. BROWNING AND ROBERT H. MELLEN. *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—Second-order acoustic theory holds that a plane sinusoidal pressure wave becomes sawtooth in shape as it propagates. Experiments showing the progressive distortion of a 150 kHz sine wave radiated from a square (40×40 cm) transducer are described. With an initial peak pressure of approximately 1 atm, the distortion becomes highly developed in a 10-m water path. However, the waveform appears to differ slightly from theory.

THURSDAY, 16 NOVEMBER 1967 MEDALLION WEST, 2:00 P.M.

Session BB. Speech IV: Intelligibility and Varia

HARRY HOLLIEN, *Chairman*

Contributed Papers (10 minutes)

BB1. Masking of Crosstalk by Speech and Noise. TAPAS K. SEN (nonmember), *Bell Telephone Laboratories, Holmdel, New Jersey 07733.*—Thirty subjects took part in two laboratory experiments designed to measure detectability and intelligibility thresholds of crosstalk (a portion of speech energy that occasionally gets transferred from one channel to another channel during telephone transmission). Experiment I involved masking of crosstalk by white noise. Experiment II involved masking of crosstalk by white noise and also primary speech. Simulated telephone conversations were used as primary speech. Short sentences were used as crosstalk. In Expt. II, ratings of transmission quality also were obtained. The threshold versus noise functions were found to be linear for high-noise conditions (4-dB pressure spectrum level and higher) and markedly nonlinear for lower noise values. Intelligibility thresholds were found to be 8–10 dB higher than corresponding detectability thresholds. Also, thresholds obtained with both background noise and speech (Expt. II) were found to be higher by 8–10 dB than comparable thresholds obtained with background noise only (Expt. I). The rating data showed that transmission quality was determined largely by the primary speech volume and noise in the circuit, depending very little on crosstalk.

BB2. Intelligibility of Individual Consonant Sounds Distorted by Infinite Peak Clipping. MAURICE E. JOSEPH (nonmember), *University of Illinois at the Medical Center, Chicago, Illinois 60612.*—Previous experiments on infinite peak clipping, using whole-word intelligibility as criterion, have shown that speech may retain high intelligibility when relative amplitude information in the waveform is eliminated. To determine whether this effect holds true for all speech sounds or whether individual phonemes are affected to a different degree, in the present experiment 10 listeners phonetically transcribed infinite-peak-clipped nonsense words tape recorded by three talkers. The test materials were based upon a corpus of six plosive and six fricative consonants, voiced and voiceless, combined with five vowels in VCV utterances. Eight syllable types containing the twelve consonants provided a controlled context with equal response alternatives for each stimulus. A differential effect of clipping was evident: dental fricatives were severely degraded in intelligibility, while plosives and alveolar fricatives tended to remain relatively unaffected. Intelligibility of voiced versus voiceless sounds was not differentially affected by clipping. Confusion matrices revealed that most errors occurred within the same voicing or manner-of-articulation category.

BB3. Further Studies of the Intelligibility of Alternated Speech. GEORGE W. HUGHES, ARTHUR S. HOUSE, DAVID P. GOLDSTEIN AND JOHN A. RUPF, *Purdue University, Lafayette, Indiana.*—Previous studies in which subjects were asked to repeat (i.e., shadow) speech that was switched alternately between the two ears have presented evidence that the perception of alternated speech is adversely effected when alternation is synchronous with syllables [A. W. F. Huggins, J. Acoust. Soc. Am. **36**, 1055–1064 (1964); G. W. Hughes *et al.*, J. Acoust. Soc. Am. **40**, 1283 (1966)]. In this study, we compare the perceptual effects of syllable-synchronous switching to those caused by alternation that occurs periodically at the average syllable rate. We also compare effects of alternation at phoneme boundaries with the effects of other conditions of alternation. No striking difference in shadowing performance between the syllable- or phoneme-locked alternation and the periodic alternation were observed, but a strong patterning of errors relative to position in the script was found for all modes of alternation. Our results indicate that script composition may be a more important factor than mode of alternation in determining shadowing behavior. [This work was supported in part by the Air Force Cambridge Research Laboratories under contract.]

BB4. Selecting an Intelligibility Test for Communication System Evaluation. CARL E. WILLIAMS AND MICHAEL H. L. HECKER, *Bolt Beranek and Newman Inc., Cambridge, Massachusetts, 02138.*—A variety of intelligibility tests are available for evaluating speech-communication systems. These tests have certain advantages and limitations that must be considered in terms of the particular requirements of a given evaluation task. Because any single test provides only some of the desirable features found among all tests, the selection of a test is often a compromise. To facilitate an optimal selection, it is suggested that four generalized requirements be rank ordered according to their importance in a given evaluation task: (a) the need to diagnose specific system shortcomings, (b) the need to discriminate among highly intelligible systems, (c) the need to use speech material that provides a representative sampling of the sounds occurring in everyday speech, and (d) the need for rapid test administration and scoring. Suggestions are made for the development of a new test that would hopefully incorporate more desirable features and thus reduce the present restrictions encountered in the selection of a test.

BB5. Automatic Evaluation of Time-Varying Communication Systems. MICHAEL H. L. HECKER, GOTTFRIED VON BISMARCK (nonmember), AND CARL E. WILLIAMS, *Bolt Beranek and Newman Inc., Cambridge, Massachusetts, 02138.*—Many types of speech-communication systems may be readily evaluated with available electronic devices which employ the concept of the Articulation Index (AI) (e.g., the Speech Communication Index Meter). To study the feasibility of using such devices to evaluate time-varying systems, recordings were made of the transmission characteristics of a troposphere scatter system. At many specific points in time in these recordings, AI's were calculated and intelligibility scores were obtained from listeners with the aid of a special test procedure. For most points, the intelligibility score could be reasonably well predicted from the AI. This finding was interpreted as indicating that the existing devices are potentially capable of evaluating time-varying systems.

BB6. Speech Sounds in Atypical Environments. JERRY D. SPEAKMAN (nonmember), *Biodynamics and Bionics Division, 6570th Aerospace Medical Research Laboratories, Wright–Pat-*

terson Air Force Base, Ohio.—Unusual breathing environments such as those containing helium are being considered for future manned space systems. In the study reported herein, the physical acoustic characteristics of speech generation were investigated for helium concentrations of 0%–70% at pressures from sea level to 258 mm Hg. An electromechanical-acoustical model of the vocal tract for the sustained phonetic vowel sounds of [i] (eat), [o] (lost), and [u] (boot) was utilized to determine the shift in formant frequency and the change in acoustic power radiated relative to that for normal air. For comparison purposes, the relative acoustic powers were calculated for the sss and th fricatives, some simple ideal acoustic generators and moving voice coil-type loudspeakers; in addition, loudspeaker directivity characteristics and power output were experimentally measured. Previously noted impaired communication capability in these environments can be attributed to speech spectrum changes due to the nonuniform power reduction of vowels and consonants and changes in formant frequency and directivity.

BB7. Speech Intelligibility as a Function of Helium-Oxygen Breathing Mixture and Ambient Pressure. HARRY HOLLIEN AND CARL L. THOMPSON, *Communication Sciences Laboratory, University of Florida, Gainesville, Florida.*—Speech intelligibility was obtained on Navy divers at sea level (in air) and at 200 and 450 ft (simulated) in helium–oxygen breathing mixtures. Subjects were personnel in training for Sealab 3 at the Experimental Diving Unit, Washington Navy Yard. A total of 26 subjects were evaluated (11 at 0 and 200 ft, 11 at 0 and 450 ft and four at 0, 200, and 450 ft). The speech material used was the PB-25 word lists equated by Campbell for difficulty; the lists were read prior to descending and *immediately* upon reaching depth. The helium–nitrogen–oxygen mixtures were 79/17/4 at 200 ft and 90/8/2 at 450 ft. A relatively nonreverberent recording environment was provided by an enclosure fabricated from mattresses available in the chamber's sleeping compartment. Listeners were 15 University of Florida students trained for tasks of this nature. The combined efforts of HeO_2 mixtures and high ambient pressures were extremely detrimental to speech intelligibility (scores of less than 20% were common at 450 ft). [The project was supported by the Office of Naval Research.]

BB8. Influence of Bandwidth and Reverberation Time upon Preference Evaluation of Speech. G. M. SESSLER AND J. E. WEST, *Bell Telephone Laboratories, Incorporated, Murray Hill, New Jersey.*—The subjective influence of bandwidth, center frequency, reverberation time and amplitude and delay of early echoes on speech was studied. Selected sentences spoken by different talkers were recorded in an anechoic chamber by a broad-band system. These sentences were reverberated on a digital computer with reverberation processes corresponding to T_{60} = 0 to 0.5 sec but having specified early echo structure. After this process, bandpass filtering in half-octave steps with bandwidths of 2–5 oct was applied to the sentences. The test stimuli so generated were presented in paired-comparison tests to a number of subjects. The subjects were asked to select that signal from each pair that they believed to be preferable for communication systems. Evaluation of the subjective results was performed on a digital computer using a factor-analysis program. It was found that of the parameters listed above bandwidth is the single most important one for assessing subjective quality. Also, subjective quality increases in general with decreasing reverberation time. This suggests that, within limits, excessive reverberation can be compensated by an increase in the bandwidth of the transmitted signal.

BB9. On Line Analysis of Subjective Preference Data. H. S. MAGNUSKI (nonmember), G. M. SESSLER, AND J. E. WEST; *Bell Telephone Laboratories, Murray Hill, New Jersey.*—The preference data described in the previous abstract was analyzed through use of an on-line graphical display. The Graphic 2 console, consisting of a small computer and associated display processor, was programmed to permit the rotation of a three dimensional space and projection of labeled points in this space onto a two-dimensional surface. The coordinate perpendicular to the display surface was represented by the size of the point. By use of a keyboard connected to the computer, the user can intensify the brightness of selected groups of points for independent study. The method permits quick evaluation of three-dimensional data. The first application of this program was the study of results relating bandwidth and reverberation time to speech quality. Results obtained by this method will be demonstrated.

BB10. Evaluation of Underwater Speech Communicators. HARRY HOLLIEN, ROBERT F. COLEMAN (nonmember), AND CARL L. THOMPSON, *Communication Sciences Laboratory, University of Florida, Gainesville, Florida.*—Evaluation of speech intelligibility—under optimum conditions—was carried out on a number of military and commercial underwater communicators. The site of the research was NRL's Underwater Sound Reference Division's Bugg Springs facility and the communicators evaluated were: (1) Aquaphone, (2) Aquasonics, (3) Bendix Watercom, (4) MAS (prototype), (5) PQC, and (6) Yack Yack (two configurations each were available for all units except Bendix and MAS). The group of talkers consisted of four Navy divers, four phoneticians and four women; however, slightly different groups were used for Bendix, PQC, and MAS. Divers descended 35 ft to DICORS and read PB-25 word lists equated for difficulty by Campbell. Recordings were made at the surface on an Ampex 601. Subsequently, intelligibility scores were obtained by playing the speech material to 12 listeners. Communicator intelligibility scores varied from levels close to zero to slightly over 50%. The poorest scores were obtained from the PQC and units utilizing throat microphones; the highest from the Aquasonics systems. [The project was supported by the Office of Naval Research.]

BB11. Vocal-Tract Cross-Sectional-Area Measurements with the Impedance Tube. P. MERMELSTEIN, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Theoretical studies indicate that the vocal-tract cross-sectional area can be calculated from the tract impedance measured at the lips. The impedance tube method yields this information by external excitation of the tract through the lips with a broad-band pulse train and analysis of the reflected signal [M. R. Schroeder, J. Acoust. Soc. Am. **41**, 1002 (1967)]. Test results for a motor driven constriction moving in a uniform tube, as well as results with one subject executing consonant–vowel articulations are presented. The results are in the form of computer-generated movies where individual frames correspond to independent area determinations from the response to excitation pulses separated in time by 10 msec. The extension of the method to nasal sounds is discussed. Impedance-tube measurements accompanied by simultaneous x-ray cineradiography are expected to determine the as yet imprecisely known transformation between sagittal view tract measurements and the corresponding cross-sectional-area values.

THURSDAY, 16 NOVEMBER 1967 SILVER CHIMES EAST, 2:00 P.M.

Session CC. Underwater Acoustics V: Reverberation and Noise

VICTOR C. ANDERSON, *Chairman*

Invited Papers (25 minutes)

CC1. A Statistical Theory of Reverberation. DAVID MIDDLETON, *Consulting Physicist, Concord, Massachusetts 01742.*—A general analytic theory of reverberation has been constructed for the common cases of a weakly turbulent medium, where surface and volume scattering are represented by collections of independent point scatterers obeying suitable Poisson laws in space [Middleton, IEEE Trans. Inform. Theory (July 1967)]. The scattering portions of each medium are described by characteristic position-dependent stochastic linear and time-varying filters, to which are assigned a hierarchy of basic structures. The result is a combined physical-phenomenological model that uses wave theory to determine the form of typical scattered waves, and ray theory to incorporate them into the governing geometry. The intractably complex scatter boundary conditions are embodied in the statistical filter. Frequency-selective apertures, general geometries, and signals are also incorporated. The theory can readily be extended to include nonzero velocity gradients, absorption, and discrete multipath phenomena. Initial experiments show good to excellent quantitative agreement (Kincaid and Rustay, General Electric Co.), including predicted volume and surface effects and spectral broadening caused by inherent doppler of the scatterers. [Work supported by the Office of Naval Research.]

Contributed Papers (10 minutes)

CC2. Reverberation Resolution Using Various Waveforms. A. DONN COBB (nonmember), *U. S. Navy Underwater Sound Laboratory, Bermuda Research Detachment, FPO New York 09560.*—This paper describes the results of an experiment conducted to determine the usefulness of various waveforms to resolve discrete reverberators. Returns from several topographic reverberators were obtained by using waveforms of essentially the same time duration. Continuous-wave (CW), hyperbolic frequency modulation, and pseudorandom noise were transmitted in an ocean basin and their reverberant returns recorded at the output of matched filters. The comparison is made on the basis of the resolving power of the waveforms used. In general, the results indicate that at short range, where many reverberant returns are received, the coded waveforms are more effective than CW. Conversely, at long range for discrete returns, it appears that CW is more effective, since the in-phase summing is less affected by the temporal and spatial stability of the wavefront.

CC3. Reverberation Received by a Hydrophone Separated From an Active Acoustic Source. JAMES H. PROUT, *Ordnance Research Laboratory. The Pennsylvania State University, State College, Pennsylvania.*—Reverberation equations have customarily been derived only for a source and receiver at the same point as in the familiar sonar situation. However, the study of interference between two similar active acoustic systems requires a knowledge of the characteristics of the reverberation received at a point separated from the source. A simplified expression for volume reverberation is derived for directional projectors oriented in the same direction but displaced along the beam axis. Reverberation decay curves are presented with source-to-receiver separation as a parameter. The variation of the volume reverberation index, J_v (usually considered a constant), is also discussed.

CC4. Born Approximation in One-Dimensional Scattering Problems. CHARLES SOODAK (nonmember) AND MARK HARRISON, *The American University, Washington, D. C. 20016.*—If one attempts to find the transmission of sound through an arbitrary scattering layer by the use of standard integral equation techniques, an unexpected difficulty with the Born approximation is encountered. The first Born approximation, i.e., the unperturbed plane wave, gives a transmission coefficient greater than unity for all scattering layers. This is clearly incorrect. In this paper, a technique is presented for resolving these difficulties. It is shown that a scattering layer, whose index of refraction varies as a function of the coordinates in a complicated fashion, can be approximated by a series of layers with constant index of refraction provided that one considers only wavelengths large in comparison to the extent of each layer. These approximations give the needed corrections to the first Born approximation. This second Born approximation is found to give correct answers for the transmission of sound. Some calculated results for typical scattering layers that might be encountered in the ocean are presented. The results are compared with exact answers obtained by partial-wave methods.

CC5. Estimate of Volume Scattering Strength from Biological Samples. FREDERIC P. FESSENDEN (nonmember), *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—Biological samples of marine organisms obtained from an area in the Western North Atlantic south of New England in August 1966 have been used to estimate the volume scattering strength of the deep scattering layer. Specimens were collected by the deep submersible STAR III on six different occasions; each of three conditions of the deep scattering layer (descending, ascending, and surface layers) was examined twice. For each case the acoustic cross section and the animal concentration were used to determine the scattering coefficient values as a function of depth, and from these results scattering strength values were obtained. Scattering strengths are presented for single frequencies of 4 kHz and 12 kHz as well as for several frequency bands, using the theory of Mohammed [(J. Acoust. Soc. Am. **41**, 177–181 (1967)] for calculating average backscattering coefficients. At 4 kHz scattering, strength values ranged from -46 to -84 dB; at 12 kHz values ranged from -50 to -59 dB. A comparison of these results with experimental values obtained in similar areas shows reasonable agreement for frequencies near 12 kHz; in general, the results near 4 kHz are lower than those of other investigations.

CC6. Volume-Scattering Observations in Puget Sound and Nearby Waters. P. H. MOOSE, *Honeywell Inc., Seattle, Washington.*—In March of 1967, observations of volume scattering at 28.5 kHz were made in Puget Sound and nearby waters. Both day and nighttime measurements were taken in eight sequences, spaced over several days, at each of four stations. Pulse widths of 1, 3, and 10 msec were transmitted, the shorter pulses having a greater ability for resolving individual scattering organisms. The observed return consisted of a continuous envelope of reverberation, plus individual echoes from single scatterers. A nighttime upward migration of the scattering layer was observed at most stations. At two stations the average volume scattering coefficient (Sv) attained a value of -46 dB (*re*: yd²/cubic yd) at a depth of 100–200 ft. The average value of target strength for the 395 individual targets observed was -40 dB, and the extreme values were -85 and -29 dB.

CC7. Volume Scattering at 3 kHz and 12 kHz off Southern California. W. E. BATZLER AND R. J. VENT, *Naval Undersea Warfare Center, San Diego, California.*—Directional sound sources aimed vertically downward and using pulses of 25 and 50 msec duration at 3 and 12 kHz were used to obtain volume scattering information in a 2000-fathom area about 200 miles southwest of San Diego. Results during a two-day period expressed as average scattering strength for a water column of unit cross section show -77 ± 2 dB at 3 kHz and -72 ± 2 dB at 12 kHz. Prominent scattering layers were measured at both frequencies during the daylight hours; that at 3 kHz maintained position and strength throughout the night while the 12-kHz layer moved toward the surface at night. These tests of December 1964 tend to agree with similar data obtained in March, November, and July, and with earlier data off Southern California at 18 and 24 kHz. Near-surface scattering was not measured during the 1964 tests, but recent nighttime tests at 3 kHz show high levels near the surface. Scattering levels in this region of high organic productivity are higher than in most of the 50 or more other areas sampled in the Eastern and Western North Pacific.

CC8. Volume Reverberation Measurements between Bermuda and Puerto Rico. R. H. ADLINGTON *The Defence Research Establishment Atlantic of the Defence Board of Canada, Dartmouth, Nova Scotia.*—Measurements of volume reverberation were made at 11 stations spaced approximately 75 miles apart on a line between Bermuda and Puerto Rico. The sound source was a 1-1b TNT charge exploded 18 in. below the surface and the receiver was an omnidirectional hydrophone suspended at 30 ft below the surface. The data have been analyzed in octave bands from 1.6 to 25.6 kHz. The octave-band scattering strength of the water column drops sharply south of Bermuda but experiences a slight recovery near Puerto Rico. These measurements are in substantial agreement with those made by other observers.

CC9. Sea Noise Statistics. ELIZABETH M. ARASE AND T. ARASE (nonmember), *Hudson Laboratories of Columbia University, Dobbs Ferry, New York 10522.*—The statistics of ambient noise in the ocean have been investigated for single receivers and a closely spaced vertical array in the frequency range from 200 to 1600 Hz in ½ oct bands. Amplitude samples were taken at 20 msec intervals and were recorded digitally. The data were analyzed in blocks of 500, 1000, and 2000 data points. Distribution functions and moments up to the fourth order were computed and the samples were tested as to their normality and stationarity. A greater percentage of the smaller samples, i.e., 500 points, fall within the 95% acceptance intervals for normality than the longer samples, thus confirming the nonstationarity of ambient noise in the ocean. [Hudson Laboratories of Columbia University Informal Documentation No. 145. This work was supported by the U. S. Office of Naval Research.]

CC10. Relationship between Deep-Water Ambient Noise and Local Wind Field Fluctuations. LLOYD C. HUFF (nonmember), *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—The results of two comprehensive independent investigations on ambient noise and wind field fluctuations versus wind speed are combined to postulate an interconnection between deep water ambient noise and the short term fluctuations about the local mean wind field. The model considers the air/sea interface to be a time changing acoustic impedance and that energy is extracted from the wind field via a turbulent boundary layer. The combined results indicate that for frequencies above 446 cps, the sound pressure level in dB is linearly related to 20 log of the short-term speed fluctuations, and that for lower frequencies down to 112 cps, the SPL approaches this relationship for mean wind speeds above 22 knots. This model affords a reasonable explanation for the observed fact that at high wind speeds, the level of ambient noise is less sensitive to changes in wind speed. Data that represent a direct evaluation of the postulate that ambient noise is physically linked to the kinetic energy of the local wind-speed fluctuations will be presented.

CC11. Some Applications of Meteorology to Underwater Ambient Noise Studies in Block Island Sound. LLOYD C. HUFF (nonmember), AND ROBERT G. WILLIAMS, *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—Turbulent atmospheric boundary-layer theory is applied to wind observations made over a shallow-water embayment to explain variations in ambient noise levels. Broad-band ambient-noise data for sea states up to 3 obtained at a fixed receiving site are presented for a shallow water acoustic test range in Block Island Sound. Hourly wind-speed averages are analyzed by means of spectra and covariance functions in order to compare the frequency composition of the acoustic and meteorological data. The power spectrum computed from the record of ambient noise pressure level as a function of time has significant peaks centered on frequencies of 0.04 and 0.10. Similar peaks at the corresponding frequencies are present in the spectra of wind speeds. The results of this experiment suggest that for wind speed fluctuations of less than 0.33 nonlinear effects of the wind are relatively unimportant in the generation of ambient noise.

THURSDAY, 16 NOVEMBER 1967 — EMPIRE ROOM, 2:00 P.M.

Session DD. Shock and Vibration II: Vibration of Plates, Shells, and Structure

D. F. MUSTER, *Chairman*

Contributed Papers (10 minutes)

DD1. Modal Energy Division in Semi-Infinite Plates. PETER J. TORVIK AND JAY J. MCCLATCHEY (nonmember), *Air Force Institute of Technology Wright-Patterson Air Force Base, Ohio 45433.*—The stress and displacements within a semi-infinite elastic plate resulting from a time-varying (sinusoidal) edge loading are determined in terms of the modes of the infinite plate. A uniform normal stress with zero shear stress was supplied at the edge. The coefficients in the expansion are determined by a method given previously [J. Acoust. Soc. Am. 41, 346–353 (1967)]. Frequencies up to the frequency where five propagating modes are present are considered and the fraction of the supplied energy going into each of the modes determined. In the frequency range considered, the dominant mode changes from the first to the second and then to the third. A method for predicting the dominant mode requiring only a knowledge of the real roots of the Rayleigh–Lamb equation is suggested.

DD2. On the Vibration of an Elastic Plate on an Elastic Foundation. DAVID H. Y. YEN (nonmember), *Michigan State University, East Lansing, Michigan* AND S. C. TANG (nonmember), *Ford Motor Company, Dearborn, Michigan.*—This paper is a study of the vibration of an elastic plate under a time-harmonic point force. The plate is infinite in extent and rests on an elastic foundation. This study is made on the basis of the improved (Timoshenko) plate theory. The mathematical problem is to seek a fundamental solution of the time-reduced plate equation of the improved plate theory. Such a fundamental solution is constructed by the distributional Fourier transform method. From the explicit expressions of the fundamental solution is examined in detail the behavior of the fundamental singularity as a function of the vibration frequency and the foundation stiffness. Also, it is found that several types of plate resonance may occur depending upon the combination of the vibration frequency and the foundation stiffness. The paper is concluded with a discussion on how to construct integral representations for steady-state vibrations (Green's functions) for finite plates. [Research supported in part by the National Science Foundation.]

DD3. Behavior of Complex Propagation Roots in Thick and Thin Cylindrical Shells. EDWARD M. FRYMOYER, *Ordnance Research Laboratory, The Pennsylvania State University, State College, Pennsylvania.*—On the basis of linear elasticity analysis, the set of modes, which may exist in an infinitely long cylindrical shell, have real, pure imaginary, and complex propagation numbers. Previous investigations, except for a recent paper by McNiven *et al.* [J. Acoust. Soc. Am. 40, 1073–1976 (1966)], have concentrated almost entirely on the real branches, but the complex and imaginary branches are important when general boundary conditions of a finite shell are considered. The complex branches of the mode dispersion curves for elastic curves in an isotropic cylindrical shell occur at lower frequencies than in a solid cylinder of the same dimensions. The modes describe the end distortion displacement component in the characteristic vibrations of a finite cylinder. The change from thin to thick shell behavior is traced. The complex branch intersects on the imaginary plane for thin shells and on the real plane for thick shells. The exact transition from thin to thick shell behavior is a function of Poisson's ratio. For a Poisson's ratio of 0.3, it occurs at a thickness/mean radius ratio of 0.65 for the axially symmetric case. The results of approximate analysis based on thin-shell theory (Flügge) and thick-shell theory (Mirsky-Herrmann) are compared to three-dimensional linear elasticity analysis. For thin shells, the approximate analysis reproduces the full low-order three-dimensional branches in detail. [This investigation was performed at the Ordnance Research Laboratory under U. S. Naval contract with the Naval Ordnance Systems Command (NOSC) and the Office of Naval Research (ONR).]

DD4. Vibration of Cellular Shells. SABIH I. HAYEK AND MANU C. PATEL (nonmember), *Department of Engineering Mechanics* AND *Ordance Research Laboratory, The Pennsylvania State University, University Park, Pennsylvania 16802.*—A method is presented for determining the natural frequencies of an elastic cellular shell of finite length. The shell is made of two concentric shells, with equally spaced ring stiffeners placed in between. For the sake of formulation, the shell is replaced by an equivalent multilayered solid-wall shell, the layers being homogeneous and anisotropic. The equations of motion are obtained by Hamilton's principle. The natural frequencies are computed for various shell geometries and mode numbers.

DD5. Response of a Conical Shell to a Turbulent Boundary Layer Pressure Field. K. L. CHANDIRAMANI (nonmember), *Bolt Beranek and Newman Inc., Cambridge, Massachusetts 02138.*—A thin truncated conical shell can be looked upon as a series of cylinders of varying diameters. This point of view, in conjunction with some well-known vibration properties of rectangular flat plates, circular flat plates, and cylinders, yields an approximate description of the vibration properties of conical shells. Comparison with available experimental results is encouraging. The same approximate approach is used in a slightly different form for estimating the vibration response of a conical shell to a turbulent boundary-layer pressure field. Response is found to increase gradually from the smaller end of the conical shell to the larger end.

DD6. Free Vibrations of Thin Orthotropic Oblate Spheroidal Shells. LESLIE E. PENZES (nonmember), *General Dynamics/Convair, San Diego, California.*—The problem of the free vibrations of a thin orthotropic, oblate spheroidal shell has been solved by the assumptions of orthotropic membrane theory and harmonic axisymmetric motion. The theory has reduced the differential equations of motion to a single ordinary second-order differential equation with variable coefficients. The resulting eigenvalue problem is solved by Galerkin's method. The principal directions of the elastic compliances are assumed to be along parallels of latitude and along meridians. Both the oblate spheroidal and spherical shells were studied by varying the orthotropic constants. The existence of finite bounds for the corresponding mode shapes has been shown. Special consideration is given to the isotropic shell as a limiting case of the orthotropic problem. The present theory in the isotropic case yields to the author's previously published results for the isotropic oblate spheroidal shell as well as the well-known exact solution of the isotropic spherical shell.

DD7. On the Response of Linear Structures to Classes of Pressure Fields. MILT G. COTTIS, *Douglas Aircraft Company, Santa Monica, California.*—With the use of a general expression for the Green's function appropriate to a homogeneous, linear structure of finite extent and constant damping, the response integral representation is transformed to the wavenumber (k) and frequency (ω) spaces. Instead of specifying particular functions for the pressure field, general analytic properties of its spectra in the complex k and ω planes are assumed and the appropriate integrations are carried out with the use of residue theory. Response solutions are obtained for the important class of pressure fields whose spectral representations have poles of any order, including essential singularities. The field itself may be random or deterministic (truncated in space, time, or both), stationary or convecting. It is shown that the general behavior and physical properties of the response to any given load can be predicted with no computation on the basis of the analytic properties of the field in the complex k and ω planes. Application of this new formalism to a particular class of fields and to the question of pressure field simulation is discussed.

DD8. Shape Factors in Low-Frequency Noise Reduction. J. RONALD BAILEY (nonmember) AND FRANKLIN D. HART, *Department of Mechanical and Aerospace Engineering, Raleigh, North Carolina 27607.*—An analysis is made of noise reduction of small enclosures in the low-frequency range where both panel and volume are stiffness controlled. The strong dependence on geometry of the compliance of flexible panels defined in terms of volume displacement and internal–external pressure difference is illustrated. Shape factors for low-frequency noise reduction are established through expressions for the acoustic compliance of enclosures and the compliances of rectangular, circular, cylindrical, and spherical panels, and a comparison of the noise reduction of enclosures involving these elements that have equal volume, panel thickness, and exposed surface area. It is observed that the stiffer, membrane-controlled spherical and cylindrical enclosures have greater noise reduction than enclosures having flexure-controlled flat panels. A companion experimental investigation is described and the results are discussed and compared with the theory. [Work supported by NASA.]

DD9. Matrix Analysis of Linear Structures Subject to Stationary and Correlated Random Inputs. V. J. Kouskoulas, (nonmember), *Douglas Aircraft Company, Santa Monica, California* and W. C. Hurty (nonmember), *University of California, Los Angeles, California*.—On the basis of correlation and spectral theories of complex random functions, this paper presents a unified matrix theory for the response of linear time invariant structures to stationary and statistically correlated random inputs. This theory is applicable to any structure whether it is a beam plate or shell, provided that its influence coefficients may be established. Various properties of the Input–output statistical matrices are exposed and formally proved for the simplifications they introduce in practical applications. The idea that the random response of a structure need not be based on its response to some deterministic input is fully employed. Indeed, no deterministic response is assumed and the Admittance matrix is established for nonproportional and proportional damping.

DD10. Vibrations of an Oval Ring. G. A. Brigham, *Autonetics, Division of North American Aviation, Inc. Anaheim, California 92803*.—Love's equations of a curved rod in flexure are applied to an oval ring having symmetry about an X axis and a Y axis. The original set of equations is then reduced by successive approximations to one relating the normal deflection and the applied forces. For an oval shape similar to that used by Marguerre and later by Romano and Kempner, the deflection may be expressed as a series of Mathieu functions, which are the normal modes. An ellipse may be approximated by this method if the ratio of minor to major axis $\geq\frac{2}{3}$. Comparison is made between the predicted and measured deflections and natural frequencies of a 1-in. thick aluminum oval module of axis ratio $\doteq\frac{1}{3}$ which are commonly used in the construction of flextensional transducers. The model is restricted to an eccentricity parameter less than unity.

DD11. Additive Property of Modal Density for a Composite Structure. F. D. Hart and V. D. Desai, *Department of Mechanical and Aer space Engineering, North Carolina State University, Raleigh, North Carolina 27607*.—It has been postulated that the modal density for a composite structure is equivalent to sum of the modal densities of the idealized substructures. A demonstration of the validity of this postulate for a particular composite is given by analyzing a structure consisting of two beams joined at right angles to form an L-shaped frame. The frequency equation for the composite and its asymptotic representation are utilized to determine the cumulative number of modes for the structure as a function of frequency. Modal density is determined from a graphical presentation of these results and compared with the sum of the modal densities of the two substructures. Consistent agreement is obtained for all frequencies giving rise to the additive property of modal density for the structure analyzed. [Work supported by NASA.]

Thursday, 16 November 1967 — Medallion East and West, 8:00 p.m.

Session WE. Engineering Acoustics Workshop: Formulation and Solution of Acoustic Radiation Problems as Integral Equations

Harry A. Schenk, *Chairman*

Invited Papers (20 minutes)

There has been considerable recent interest in the formulation and solution of acoustic radiation and diffraction problems as integral equations. This effort has been spurred largely by the advent of large-scale digital computers that make practical the approximate, but accurate, solution of these integral equations. Some of this effort has been reported in recent publications, but new results and increased understanding make a treatment of this subject timely. In this Workshop, some fundamental aspects of this topic will be reviewed and new material presented. There will be opportunity for discussion of each paper and a liberal amount of time after the papers have been presented for pertinent comments and criticism from the audience. A bibliography on the subject of this Workshop will be distributed.

WE1. Existence and Uniqueness of Solutions of Integral Equations for Acoustic Radiation and Diffraction Problems. Ben Noble (nonmember), *U. S. Army Mathematics Research Center, University of Wisconsin, Madison, Wisconsin*.—Although a radiation or diffraction problem may have a unique solution, certain formulations of the problem may result in integral equations that do not have solutions, or whose solutions are not unique. Typically, when considering the exterior problem, trouble may occur at frequencies corresponding to resonance in the interior of the body. This paper summarizes our present state of knowledge concerning existence and uniqueness. The implications in connection with approximate numerical solution of the integral equations are examined. Methods are discussed for obtaining integral equations that always have unique solutions. The theoretical advantages and disadvantages of various methods will be compared.

WE2. Calculation of Acoustic Fields about Arbitrary Three-Dimensional Bodies. John L. Hess (nonmember), *McDonnell–Douglas Corporation, Long Beach, California*.—A general method is described for calculating radiated and scattered sound fields about three-dimensional bodies. The method utilizes a source distribution on the surface of the body and solves for the distribution necessary to satisfy the boundary conditions. Plane quadrilateral surface elements are used to approximate the body surface, and the integral equation for the source density is replaced by a set of linear algebraic equations for the values of source density on the quadrilateral elements. Once the source density is known, acoustic pressure and fluid velocity on the body, in the nearfield, and in the farfield are calculated. The acoustic fields of the quadrilateral elements are evaluated by certain expansions in wavenumber. The method is described and its usefulness and limitations discussed. The accuracy of the method is exhibited by comparisons with analytic solutions for a number of bodies including a sphere, a cube, a triaxial ellipsoid, and a finite circular cylinder.

WE3. Operator Singularities of Integral Equations of Linear Acoustics. J. L. Baylor (nonmember), *Electric Boat Division of General Dynamics, Groton, Connecticut.*—The harmonic potential which represents the field radiated from a closed surface is a solution of the reduced wave equation. On the radiating surface, the potential and its normal derivative are uniquely related by a surface integral equation. This integral equation is singular in the operator sense at the eigenvalues of the reduced wave equation for the zero potential surface condition. The way in which these operator singularities influence the analytical and numerical solution of the integral equation will be discussed. Analytical and numerical examples which illustrate the eigenvalue influence will be given.

WE4. Approximate Numerical Representation of the Helmholtz Integral Operator. R. F. Pohler (nonmember), *TRACOR, Inc., Austin, Texas.*—The discussion is concerned with numerical aspects of solving the Helmholtz integral equation of acoustics. The various numerical techniques which are used are interpreted in terms of an abstract vector space. From this point of view, the system of algebraic equations which approximates the integral equation corresponds to a representation of the integral operator in terms of a set of basis functions which spans part of the vector space. The special properties of the integral operator for separable geometries are reviewed, and the rôle of the associated eigenfunctions as a basis set is discussed. The influence of the given boundary conditions on the choice of the basis set is considered. Numerical examples which compare results using different basis functions are given.

Friday, 17 November 1967 — Medallion West, 9:00 a.m.

Session EE. Shock and Vibration III

E. L. Hixon, *Chairman*

Contributed Papers (10 minutes)

EE1. Comparison of Estimated and Measured LM Acoustically Induced Vibration. M. Bernstein and L. Pulgrano, *Grumman Aircraft Engineering Corporation.*—The launch vibration levels induced during an acoustical test of the Lunar Module on several items are compared with estimates made by extrapolating statistically analyzed measured data. Procedures and data adapted from work by Mahaffey and Smith and by Barrett were used. Both methods were generally conservative. Estimates for a densely arranged $\frac{1}{2}$ ton electronic equipment rack were particularly high. The use of component and support structure weights as attenuation factors is described to demonstrate the necessarily large reliance on engineering judgment in these procedures. The acoustical test, in which a representation of the entire Apollo system is excited through contiguous external ducts, is described briefly. An envelope of measurements of 282 accelerometer locations on the ascent stage considered as measuring equipment response is compared with 75 locations considered "primary" structure motion to demonstrate that specifying an envelope over the latter can be very conservative.

EE2. Mechanical Vibration Transmission in the Mariner '69 Spacecraft. Terry D. Scharton (nonmember), Thomas M. Yang (nonmember), *Bolt Beranek and Newman Inc., Van Nuys, California 91406* Marc C. Trummel, and Momtaz N. Mansour (nonmember) *Jet Propulsion Laboratory, Pasadena, California 91103.*—Statistical energy techniques are used to obtain an engineering estimate of the high-frequency mechanical vibration transmission in the Mariner '69 spacecraft assembly. The vibration transmission path involves the following elements: (1) the Cenatur tank dome, or alternatively, a vibration test fixture, (2) the spacecraft adapter, and (3) the spacecraft electronic assemblies. Analytical values of the acceleration transfer functions for these elements are compared with measured values obtained in development vibration tests of a structural-model spacecraft. The results of this comparison are used to refine the analysis techniques and to gain confidence for applying the techniques to other spacecraft structures. Plans to study the acoustical vibration transmission in the Mariner '69 are described.

EE3. Acoustical Qualification of Saturn S-IC Fin Structures. David R. Kennedy, *Brown Engineering Company, Huntsville, Alabama* and Clark J. Beck (nonmember), *Boeing Company, Huntsville, Alabama.*—The four aerodynamic fins, which are located close to the F-1 engines on the S-IC stage of the Saturn V vehicle, will be subjected to very high noise levels during both launch and flight. High-level acoustic energy is recognized as a potential cause of structural failure; hence, acoustic qualification tests were carried out on four fin specimens in a progressive wave chamber. Under identical test conditions (163 dB for 20 min), two of the specimens failed while two survived with no fatigue cracks detectable. This difference between the test results is attributed to the use of a thin layer of fiber glass tape in the construction of the two fins that did not fail. This tape was primarily used to separate the dissimilar metals of the ribs and skin. This paper describes the tests and presents data which show the reduction in strain levels produced by this thin layer of tape. [This work has been supported by the National Aeronautics and Space Administration.]

EE4. A Broad-Band Versus Narrow-Band Measure of Spectral Peaks in Vibration Data. Marc C. Trummel and David B. Wiksten (nonmember), *Jet Propulsion Laboratory 4800 Oak Grove Drive Pasadena, California 91103.*—Spacecraft vibration caused by wide-band excitations may be subdivided into structurally related zones and presented a space average spectrum levels percentile spectrum levels, maximum envelopes, etc. In many cases, it is desirable or actually required to use frequency smoothed data such as the ⅓-oct band format. This requirement arises when predicting vibration levels on new spacecraft or structures by extrapolation from previous data or by analytic methods. Other applications occur in certain acoustic testing or case where only ⅓-oct data is available. As a final step in using broad-band frequency data, a peaking factor is required to estimate the narrow-band peaks to be expected. Empirical estimates of peaking factors are derived by analysis of vibration data from tests of two structurally dissimilar spacecraft: the MARINER and SURVEYOR. Data are analyzed and manipulated in ⅓-oct and narrow-band formats. Peaking factors are derived for: (1) a direct comparison of ⅓-oct and narrow-band data, (2) space averages of both types of data and, 3) 95th percentiles from sets of both types of data.

EE5. Distribution of Damage in Random Fatigue. L. PULGRANO AND M. BERNSTEIN, *Grumman Aircraft Engineering Corporation.*—The rate at which Palmgren–Miner damage accumulates under narrow-band random stress oscillations is well known. Not as well known, however, is the portion of the total damage that is attributable to peaks lying within a given stress range. It is of interest to know whether it is the relatively infrequent high-level stress peaks or the much more frequent peaks at low levels that contributes most to the total damage. A simple extension of previous work permits the development of a damage distribution function, which gives the relative contribution of peaks in any stress range to the total damage. The results indicate that most of the damage is produced by peaks lying in a stress band that is about 2σ wide. For 2024-T3 aluminum, which is considered as a particular case, the damage-producing band is centered on about 2.7σ and it contains about 23% of the total number of peaks.

EE6. SIC Rocket-Noise Effect on House Trailers. G. NORMAN SAWYER, *General Electric Company* AND L. W. NYBO (nonmember), NASA *Mississippi Test Facility, Bay Saint Louis, Mississippi,*—Criterion for noise-induced damage presently in use places major emphasis on window-glass breakage. The criterion sets noise levels at the frequency of the window resonance considering the window to be simply supported. In the case of light trailer construction, the effect of the wall driving the window should be considered. Measurements made on a house trailer exposed to the noise of the SATURN V's first stage booster, the 7 500 000 pound thrust SIC, static tested at NASA's Mississippi Test Facility, are presented.

EE7. Shock Expansion Tube as a Potentially Useful Tool for Sonic Boom Simulation, H. E. DAHLKE, G. T. KANTARGES (nonmember), AND J. J. VAN HOUTEN, *LTV Research Center Western Division, Ling-Temco-Vought, Inc., Anaheim, California 92800.*—Considering the emphasis being placed on studies of community reaction to sonic-boom phenomena, the simulation of these pressure transients has increased in economic and scientific importance. The expansion of shock waves produced by burst diaphragm techniques similar to those used in the shock tube provides waveforms that approximates the subjective pressure experienced during the onset of a sonic boom. Conventional shock-tube analysis, combined with Whitham's theory concerning the gas dynamics of a nonuniform channel, provides the basis for predicting the pressure time history of the waves generated in the expansion tube. Two shock tubes in a unique arrangement are attached to one acoustic horn. When the two burst diaphragms are subsequently punctured by an electrodynamically driven plunger, a pressure signature is generated which closely resembles the waveform subjectively experienced by an observer of a sonic boom. Experimental data are presented indicating the wide range of pressure amplitudes and pulse durations achievable utilizing this tool. [Work supported by the National Aeronautics and Space Administration, Langley Research Center.]

EE8. Use of Acceleration Feedback in Active Shock and Vibration Isolation. PETER C. CALCATERRA (nonmember), AND DALE W. SCHUBERT (nonmember), *Barry Control, Watertown, Massachusetts,* FREDERICK E. EZEKIEL (nonmember), *F. C. Ezekiel Company, Lexington, Massachusetts.*—This paper discusses the theoretical, as well as the practical aspects of using acceleration feedback in conjunction with an electrohydraulic closed loop control system to achieve shock and vibration isolation. The payload to be isolated is mounted on a hydraulic cylinder which is driven by a servo-valve. The signal from an accelerometer mounted on the object serves as the principal controlling signal to the servo loop. By striving to maintain zero acceleration the payload is thus isolated from disturbing base motions. A position feedback signal maintains the payload at fixed position within the cylinder even in the presence of steady base acceleration. The advantage of such a system is to provide isolation at frequencies below 1 Hz with minimal space and deflection requirements. In addition to rejecting base disturbances, the system is nearly insensitive to external forces applied to the isolated payload. A laboratory test model was designed and built to provide isolation against vertical dynamic excitations experienced by pilots during low-altitude high-speed flights. Analytical and experimental results are shown for the response of the model in the frequency and time domains.

EE9. Use of Constant-Bandwidth Narrow-Band Frequency Analysis to Pinpoint Noise Sources in Machinery. G. A. LYNCH AND L. D. MITCHELL, *Engineering Technology Laboratory E. I. du Pont de Nemours & Co., Inc., Wilmington, Delaware. 19800.*—This paper compares the results of the analysis of noise and vibration in machinery by constant-bandwidth filters as narrow as 2 cps with the more conventional octave-band, ⅓-oct band, and percentage-of-center-frequency filters. The nature and character of the resulting noise spectra are used to pinpoint machinery noise sources. The technique described involves instrumentation-quality tape recording of noise emanating from a machine, followed by analysis of the tape in the laboratory with narrow-band scanning filters and direct plotting of the results on $X-Y$ plotters. In some studies, the interdependence of noise and vibration has been determined by simultaneously recording both signals and making an analysis. Comparisons between these narrow-band vibration and noise spectra give new insight into the dynamic behavior of machines.

EE10. A Method of Axial Testing for Roller Bearing Noise Qualities. R. F. LUCHT (nonmember), *Research Division, Timken Roller Bearing Company, Canton, Ohio* AND R. H. SCANLAN, *School of Engineering and Applied Science, Princeton, New Jersey.*—A number of investigators have studied the vibration of rolling element bearings by taking samples of bearing race vibratory velocity, acceleration, or displacement at one point on the race the bearing being run in a somewhat isolated enviroment. While such methods have lead to considerable progress in the evaluation and amelioration of bearing-generated noise, there has been a constant trend toward improvement of the methods. One promising step in this process is reported here. The present paper discusses the replacement of accelerometer radial testing by accelerometer axial testing, the new connecting link between bearing and accelerometer being a metallic exponential horn. The horn technique avoids problems of accelerometer strain sensitivity and dependence of results upon accelerometer angular orientation with respect to specific bearing anomalies.

EE11. Establishment of Objective Criteria Reflecting Subjective Response to Roller Bearing Noise. R. F. LUCHT (nonmember), *Research Division, Timken Roller Bearing Company, Canton, Ohio* AND R. H. SCANLAN, *School of Engineering and Applied Science, Princeton University, Princeton, New Jersey.*—In recent years, the question of noise generated by rolling element bearings has been given considerable attention. That a certain irreducible "minimum" noise will be created by a properly functioning bearing has been pointed out in earlier work. Acceptable roller-bearing noise levels above this minimum are regarded, in this paper, as being of necessity set by subjective standards. The paper addresses itself first to the validity or consistency of certain subjective evaluations of bearing noise made by a group of observers in representative circumstances. Next, the question is examined of what quantitative objective measurements are to be made in order accurately to reflect the character of the subjective evaluations. Finally, correlations between objective and subjective data are shown to be good. A representative method is thus defined for providing objective criteria that reflect subjective appraisal in this type of problem.

FRIDAY, 17 NOVEMBER 1967 BURGUNDY EAST, 9:00 A.M.

Session FF. Psychological and Physiological Acoustics VII: Single Units; Data and Models

DONALD C. TEAS, *Chairman*

Contributed Papers (10 minutes)

FF1. Responses of Single Fibers in the Auditory Nerve of the Squirrel Monkey to Paired Phase-Locked Low-Frequency Tones. JOHN F. BRUGGE (nonmember), DAVID J. ANDERSON (nonmember), JOSEPH E. HIND, AND JERZY E. ROSE (nonmember), *Laboratory of Neurophysiology, University of Wisconsin, Madison, Wisconsin.*—Data were analyzed on-line and from analog tape using a LINC computer. Stimuli were 10-sec complex sounds generated by summing two phase-locked low-frequency sinusoids. Both frequencies were within the response area of the unit and were related in ratios of small integers. Periodograms, i.e., distributions of unit responses over one period of the complex waveform summed for all repetitions of the waveform stimulus, were studied. The findings suggest that units discharge preferentially at times when the displacements in one direction of the cochlear partition are at or near maximal values. When intensity or phase of either stimulus component is varied with a resulting change in the complex waveform, there is a corresponding change in the periodogram. Concurrently, interspike intervals are grouped around values which are integral multiples of the time between the peaks of the periodogram, and the frequency of occurrence of such intervals is a function of the amplitude of these peaks (NB-06225).

FF2. Discharge Rates of Auditory-Nerve Fibers in Response to Electric and Acoustic Stimuli. E. C. MOXON (nonmember), AND N. Y. S. KIANG, *Center for Communication Sciences, Research Laboartory of Electronics, Massachusetts Institute of Technology, and the Eaton-Peabody Laboratory of Auditory Physiology, Massachusetts Eye and Ear Infirmary.*—Responses of single auditory-nerve fibers in anesthetized cats were studied using electric and acoustic stimuli. Shocks a few decibels above threshold delivered through stimulating electrodes placed on or in the cochlea elicit responses with latencies that are both short (0.5 msec) and stable as compared with responses to acoustic clicks. For sufficiently high shock levels a one-to-one ratio of spikes to shocks can be maintained for several minutes at rates as high as 500/sec. The responses of single fibers to tones at the characteristic frequency (CF) cannot be maintained at rates higher than about 200/sec even at maximally effective stimulus levels. These results indicate that the maximum maintained rate of discharge under acoustic stimulation is not limited by the capabilities of the axon. [Work supported by NIH grants, the Joint Service Electronics Program, and a NASA grant.]

FF3. Sensitivity of Auditory-Nerve Fibers to Tonal Stimuli. N. Y. S. KIANG, E. M. MARR, (nonmember) AND D. DEMONT (nonmember), *Center for Communication Sciences, Research Laboratory of Electronics, Massachusetts Institute of Technology, and the Eaton-Peabody Laboratory of Auditory Physiology, Massachusetts Eye and Ear Infirmary.*—Spike discharges were recorded from auditory-nerve fibers in anesthetized cats. "Thresholds" for tone bursts at the characteristic frequency (CF) were determined by audiovisual detection of spikes synchronized to the bursts. "Thresholds" for continuous tones were determined from the spike rate versus stimulus level function. Both sets of thresholds give comparable graphs of threshold versus CF. When thresholds are expressed as peak-to-peak stapes displacement the points roughly describe a curve that declines 40 dB per decade of frequency. Thus, fibers with low CF tend to be less sensitive than fibers with higher CF's. For individual cats, the spread of thresholds for units with similar CF is approximately 20 dB. These findings have implications for models of excitatory mechanisms in the cochlea. In addition, they fail to support the concept that (aside from frequency-dependent differences) separate populations of auditory-nerve fibers exist with large threshold differences. [Work supported by NIH grants, the Joint Services Electronics Program, and a NASA grant.]

FF4. Peripheral Inhibition in Frog Primary Auditory Fibers. HAROLD J. LIFF (nonmember), *Johns Hopkins University, Baltimore, Maryland 21218.*—The responses of complex units in frog eighth nerve can be inhibited by adding an inhibitory tone to the excitatory stimulus. The best inhibitory frequency is greater than the unit's characteristic frequency, and outside of its tuning curve. Units do not adapt to an excitatory stimulus during inhibition and if previously adapted they show a rebound in firing rate following termination of the inhibitory tone. The effect of an inhibitory tone is as strong after several minutes as at its onset. The dynamic range of rate of firing is not reduced by adding an inhibitory tone. Tones in a certain frequency range can excite a unit while simultaneously inhibiting its response to a second stimulus. This excitation appears to be independent of whether or not the second stimulus is present [Work supported in part by the Air Force Office of Scientific Research under contract AF 49(638), and in part by the Public Health Service, U. S. Department of Health, Education, and Welfare.]

FF5. Simulation of the Neural Response in the Peripheral Auditory System. H. SAKAI AND K. OHGUSHI (nonmember), *Broadcasting Science Research Laboratories of Japan Broadcasting Corp. Tokyo, Japan.*—Response area, inhibitory area, and the phenomenon of masking in the peripheral auditory system was simulated by computer model of the basilar membrane and the nervous connection. Assuming that the primary neuron provides the forward lateral inhibition and that the input signal to the primary neuron changes in proportion to the amplitude of the vibration of the basilar membrane, the internal potential of the neuron, $n(x,f,I)$, is given by $n(x,f,I)=A\cdot g(x,f,I)-\sum A\cdot g(x+N\Delta x,f,I)\cdot w_i(N\Delta x)$, where w_i represents the distribution of the inhibition along x. If the threshold of the primary neuron is θ, the response area of the neuron corresponding to the distance x_0 will be expressed by $I=\theta/A\cdot\{g(x_0,f,1)-\sum g(x_0+N\Delta z,f,1)\cdot w_i(N\Delta x)\}$. In the presence of the second stimulus, similarly, the inhibitory area of the neuron of which characteristic frequency is f_1 will be represented by $I_2(f_2)=\{I_1\cdot n(x_0,f_1,1)-\theta\}/n(x_0,f_2,1)$. The computed results for these models agreed with the observations on the physiological experiments. In the case of masking, the intensity of the masked signal needed to exceed the threshold of the neuron corresponding to x_0 will be represented by $I_1=\{\theta-I_2\cdot n(x_0,f_2,1)\}/n(x_0,f_1,1)$. This expresses the masking pattern in the primary neurons.

FF6. Spontaneous Firing Characteristics of a Computer Model of the Peripheral Auditory System. ROBERT C. LUMMIS AND ELIZABETH A. LUNDRY, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey 07971.*—A model of the peripheral auditory system was programmed on a digital computer. The components of the model simulate the mechanical response of the middle ear and basilar membrane, and the hypothesized behavior of the hair cell and the first order neuron. In the model, the hair cell releases excitatory quanta at random times at an average rate that is a function of basilar membrane deflection. These quanta produce excitatory postynaptic potentials that add linearly in the neuron and cause firing when a threshold is exceeded. The threshold varies to simulate refractory effects. A number of parameters in the model were varied and the statistics of the resulting spontaneous firing were compared with physiological data published by Kiang *et al.* The choice of parameter values is most severely restricted by the requirement that the mode of the firing interval histogram be independent of the firing rate.

FF7. A Model for Auditory-Nerve Discharge Patterns of Mammals. DAVID J. ANDERSON (nonmember) AND C. DANIEL GEISLER (nonmember), *Laboratory of Neurophysiology, University of Wisconsin, Madison, Wisconsin.*—A model for the mammalian peripheral auditory system was implemented on the hybrid-computer facility of the University of Wisconsin. The important assumptions used are: (1) Fine terminal fibers from a short distance along the cochlear partition merge into a single auditory-nerve fiber. (2) Information is transmitted along each of the many terminal fibers in a discrete format. (3) Each terminal process recovers slowly after excitation. (4) The auditory-nerve fiber generates an output when only a small fraction of the peripheral fibers are active simultaneously except for a short refractory period when no output is possible. The model also has the characteristic of not requiring the addition of random noise to its excitatory process for the proper reproduction of data which has been most usefully observed in a statistical format. The model simulates single-spike data obtained from certain auditory-nerve fibers of Squirrel Monkey for low-frequency tone stimulation. Interval and folded histograms that have been used to describe data obtained by Rose *et al.* are matched as well as the spike-count versus stimulus-intensity relationships. [Work supported by NIH Predoctoral Fellowship.]

FF8. Discharge Characteristics of Binaurally Excited Neurons in Cat Inferior Colliculus C. D. GEISLER (nonmember), W. S. RHODE (nonmember), AND D. W. HAZELTON (nonmember), *Laboratory of Neurophysiology, University of Wisconsin, Madison, Wisconsin.*—Single neurons in the barbiturized cat's inferior colliculus responding to binaural acoustic stimuli were monitored by platinum-iridium microelectrodes. Tones and bandpass noise were used as stimuli. Usually, the stimuli to each ear had identical waveforms but differing intensities, and one stimulus was delayed relative to the other. Results confirm and extend previous observations reported from this laboratory. Characteristics of responses obtained from different neurons differed so much that it was difficult to formulate response categories. Nevertheless, certain neurons were found to respond primarily to the intensity properties of the stimuli, while other neurons responded primarily to interaural time differences. In either cases, responses elicited by tones and by bands of noise had interesting differences: it was not possible to predict the characteristics of the responses to bandpass noise from the continuous-tone responses and vice versa. [This work was supported by NIH grant.]

FF9. Binaural Interaction in the Cat Superior Olive *S* Segment. JAMES C. BOUDREAU AND CHIYEKO TSUCHITANI (nonmember), *Veterans Administration Hospital, Leech Farm Road, and Department of Pharmacology, University of Pittsburgh Medical School, Pittsburgh, Pennsylvania.*—With the exception of low-frequency characteristic frequency (CF) units, the ipsilaterally evoked discharge of the majority of *S*-segment cells can be inhibited with simultaneous stimulation of the contralateral ear. The contralateral inhibitory CF is usually identical to the ipsilateral excitatory CF. The width of the contralateral inhibitory tuning curve is similar to the width of the ipsilateral excitatory tuning curve. Partial inhibition of the ipsilaterally evoked discharge increases the interspike intervals and increases the variability of the discharge. Inhibition, like excitation, is tonically acting. The contralateral inhibitory input with equally intense tones (measured in terms of decibels above threshold) is either about as effective as the excitatory input or less effective, rarely more effective. The primary factor determining the spike output from *S*-segment cells is not the absolute intensity level of the stimulus but rather the relative intensity difference of the stimulus at the two ears. [Work supported in part by National Institutes of Health, U. S. Department of Health, Education, and Welfare.]

FF10. Structural Organization of Response Properties of Single Units in Primary Auditory Cortex of Cats. M. ABELES (nonmember), R. L. DALY, M. H. GOLDSTEIN, JR., AND J. S. MCINTOSH (nonmember), *Johns Hopkins Medical School, Baltimore, Maryland 21205.*—Single unit data from AI of immobilized unanesthetized cats were analyzed to investigate correlations between gross aspects of the arrangement of cells and coding of simple acoustic stimuli (clicks, noise bursts, tone bursts from 500 Hz to 50 kHz at moderate intensities). Twenty electrode tracks with 160 well-isolated units were located in histological sections. Angles between these tracks and anatomcial columns ranged between 1° and 63°. Consecutive units within each track having some specific common response property were identified as runs for that property. A runs test analysis indicated that for tone bursts and for clicks responsive units were clustered within the tracks. When runs of units with common sensitive frequency ranges were analyzed, long runs were found more frequently in the tracks closely aligned with the anatomical columns. However, there was no evidence that runs of responsiveness to tone bursts, to clicks, or to noise bursts were highly organized in the direction of the columns. [Work supported in part by the Public Health Service, U. S. Department of Health, Education and Welfare, and by the U. S. Air Force Office of Scientific Research.]

FRIDAY, 17 NOVEMBER 1967 SILVER CHIMES EAST, 9:00 A.M.

Session GG. Speech V: Speech-Physiology Measures

ROBERT L. RINGEL, *Chairman*

Contributed Papers (10 minutes)

GG1. Intraesophageal Pressure during Syllable Repetition. ROBERT E. MCGLONE, *State University of New York at Buffalo.*—Intraesophageal pressure was investigated as a possible indirect indicator of subglottic pressure in a dynamic speech situation. Five male subjects repeated the syllable /pʌ/ at rates of 1/sec and 4/sec, for 4 sec. Balloon length, diameter, position in the esophagus, and volume of air in the balloon were systematically varied. Of interest was the magnitude of change in intraesophageal pressure values that occur during any trial and the repeatability of the pressure measures in relation to the various experimental factor combinations. It was found that intraesophageal pressure values varied as lung volume changed; increasing as lung volume decreased. Also, variability between repeated trials was so large that the technique of intraesophageal pressure measurement appeared not to yield useful data. Amplitude of the pressure pulses associated with syllables repeated once per second appeared to interact with base pressure; decreasing as base pressure increased. No pressure pulses were found associated with syllables repeated at the faster rate. The data suggest that intraesophageal pressure cannot be used as an index of subglottic pressure in dynamic speech situations.

GG2. Dynamic Constraints in the Variation of Subglottal Pressure. M. ROTHENBERG, *Department of Electrical Engineering, Syracuse University, Syracuse, New York.*—Using a circuit-type model for the subglottal system similar to that originally proposed by van den Berg [Acta Physiol. Pharm. Neerl. 9, 361–385 (1960)], it is shown that the physical dynamic constraints of the subglottal system do not significantly limit the ability of the respiratory musculature to produce changes in subglottal pressure. The primary limitations appear to come from the dynamic limitations inherent in the mechanism of muscle contraction, with the physical system between the muscles and the pressure in the trachea adding a delay of about 15 msec. The model discussed can also be used to predict the variations of subglottal pressure resulting from changes in glottal–supraglottal air flow resistance.

GG3. Glottal Wave Periods in VCV Environments. RONALD W. WENDAHL, *Communication Sciences Laboratory, University of Houston, Houston, Texas,* AND L. DENNIS PAGE (nonmember), *Communication Sciences Laboratory, University of Minnesota, Minneapolis, Minnesota.*—Current interest has centered on the effects of vocal tract impedance changes upon glottal function. The present study was designed to determine fundamental frequency changes between vowels and consonants in simple VCV syllables. Thirteen vocally normal male subjects were asked to phonate the syllables /ava/ and /ivi/ at two fundamental frequencies. Their phonations were analyzed by visicorder–caliper methods. The results demonstrated that the consonant in each series typically and significantly lowered the fundamental frequency from 4% to 15%. Although the results were confounded in some subjects by inflectional and erratic behavior beyond our calibrated 0.2% measurement error, the results substantiated the reports of other investigators regarding the importance of mechanical coupling between glottis and vocal cavities.

GG4. Kinematic Considerations for Deriving Laryngeal Cartilage Motions from Laryngoscopic Photographs. DAVID J. BROAD, *Speech Communications Research Laboratory, Inc., Santa Barbara, California.*—The motions of the laryngeal cartilages, though taking place in three dimensions, are sufficiently constrained by the laryngeal joints to permit their derivation from two-dimensional projections such as those provided by laryngoscopic photographs. To accomplish this, equations relating the degrees of freedom of the cartilage system to dimensions measurable in the laryngoscopic view are developed and solved. The degrees of freedom are taken to be the tilt of the thyroid cartilage, the rotation of the cricoid cartilage relative to the thyroid, the rocking motion of the arytenoid cartilages on the cricoid, and the gliding motion of the arytenoid cartilages on the cricoid. The measurable dimensions selected are the medial–lateral distance between the vocal process and apex of the arytenoid cartilage, the lateral displacement of the apex, the anterior–posterior distance between the vocal process and apex, and the length of the membranous part of the local folds. The selection of these dimensions in this same order facilitates a simple solution to the equations. To implement the results still requires the estimation of certain anatomical constants.

GG5. Observation of the Larynx by a Fiberscope Inserted through the Nose. MASAYUKI SAWASHIMA (nonmember), HAJIME HIROSE (nonmember), AND OSAMU FUJIMURA, *Research Institute of Logopedics and Phoniatrics, Faculty of Medicine, University of Tokyo, Hongo, Tokyo, Japan.*—A specially devised fiberscope by use of a thin flexible fiber-optics cable was inserted through the nose in order to observe the laryngeal conditions in the course of unperturbed speech utterances. Glottal conditions during various phonatory and articulatory activities have been visually inspected and photographed. The flexible optical cable with its tip is 5.6 mm in diameter. The hard tip, 15 mm in length, houses inside an objective lens designed for a straightforward view. Glass fibers of 15 μ are used for an image guide, and the size of the image on the photographic film is 5.6×5.6 mm^2. The insertion of the optical cable does not cause any discomfort unless the nasal cavity of the subject is particularly narrow. When the scope reaches near the level of the epiglottis, the larynx is readily visible and the scope can be kept in position with a good stability. Cinefilm of the larynx obtained by this fiberscope with simultaneous sound recording will be shown at the meeting.

GG6. An Experimental Investigation of Pitch Change in Speech. JOHN OHALA, *Department of Linguistics, University of California, Los Angeles,* AND MINORU HIRANO (nonmember), *Institute of Laryngology and Voice Disorders* and *University of Southern California, Los Angeles, California.*—Electromyographic recordings of the activity during speech of selected intrinsic and extrinsic laryngeal muscles were made using three American English-speaking subjects. On all three subjects, the activity of the lateral crico-arytenoid, the cricothyroid and the sterno-hyoid muscles were sampled; on one of the three, records of the vocalis and interarytenoid muscles were also obtained plus direct recordings of the subglottal pressure. The data provided no confirmation of a recent hypothesis by Lieberman [*Intonation, Perception, and Language* (MIT Press Cambridge, Mass., 1967)] that, for American English, during other than YES–NO questions, the laryngeal tension remains relatively steady and that pitch variation is a function of the subglottal air pressure. On the contrary, the muscles studied participated actively in pitch control, and variations in subglottal pressure could account for but a fraction of the observed pitch changes. The crico-thyroid and lateral crico-arytenoid assist in raising pitch, and the sterno-hyoid assists in lowering pitch. [Work supported by National Institutes of Health, U. S. Department of Health, Education, and Welfare, and the Office of Naval Research.]

GG7. Application of Some Acoustic Measures for the Evaluation of Laryngeal Dysfunction. YASUO KOIKE (nonmember), *Institute of Laryngology and Voice Disorders, Los Angeles, California* 90025.—Perturbation of pitch period and fluctuation amplitude of voice were studied both on normal subjects and on patients with laryngeal diseases. Sound was extracted during vowel phonation by a contact micropohone through the skin and pretracheal tissues. Special attention was paid to the initial period of phonation; the data from this period were compared with those of steady periods of phonation. The investigations indicate marked perturbation of pitch and considerable fluctuation of amplitude exist in the initial period. These findings are closely related to the type of initiation. Fundamental frequency also affected the pitch perturbation and amplitude fluctuation. Pathologic cases showed different values according to the degree and nature of the laryngeal involvement in the disease process. The potential and limitatation of applying these measures for screening and evaluating laryngeal disorders are discussed.

GG8. Scaling Articulatory Behavior: Intraoral Air Pressure. ROBERT L. RINGEL, ARTHUR S. HOUSE, AND ALLEN A. MONTGOMERY, *Purdue University, Lafayette, Indiana.*—A talker's ability to evaluate certain aspects of his articulatory behavior was examined with magnitude-estimation methods. The task was to generate patterns of activity appropriate to the production of bilabial stop consonants, both in syllable initial (hold plus plosion) and syllable final (hold only) position, as well as some analogous nonspeech activities. During all activities, a measurement of air pressure in the oral cavity was obtained. The subjects monitored their production both in terms of pressure in the oral cavity, and degree of effort used in production. The findings indicate that subjects' estimation of the subjective magnitude of "pressure," or "effect," increases approximately as the 1.4 power of the intraoral pressure, a finding similar to other reported production magnitude functions. The absolute pressures used by the subjects at their "standard" levels in the plosion mode approximated values reported for stop consonant production, with higher pressures for /p/ than for /b/. In the unexploded mode of production, pressures tended to be higher, with /b/ pressures exceeding those used to produce /p/. [This study was supported in part by the Air Force Cambridge Research Laboratories under contract; and by a NIDR Research Career Development Award (RLR)].

GG9. A Cineradiographic Study of Aspiration. CHIN-WU KIM (nonmember), *Massachusetts Institute of Technology* and *University of Illinois.*—The term *aspiration* has been variously defined as a "puff of air," a glottal constriction," "a heightened glottal pressure," "a voicing lag," etc. In this study, an analysis of a cineradiographic film (anterior views of the larynx) of Korean, which is said to have three degrees of distinctive aspiration, is presented, and it will be argued that aspiration is controlled by the intrinsic laryngeal muscles, and that it is a function of a glottal opening at the time of release of a stop, i.e., the degree of aspiration is directly proportional to the minimum time that is required for the glottis to close for phonation of the following vocalic segment. It will be claimed that several phonetic phenomena, such as the aspiration of a stop before h (e.g., $t+h \rightarrow t^h$), neutralization of final stops (as in German, Korean), and the absence of aspiration after s in English, are explainable in the light of this new definition of aspiration. [This work was supported in part by U. S. Air Force Cambridge Research Laboratories under contract and by the Joint Services Electronics Program.]

GG10. Electronic Measurement of Nasality: A Study of Normal-Speaking Adults. JACK F. BENSEN (nonmember) AND FRAZER D. WHITE (nonmember), *South Florida Cleft Palate Clinic, Variety Children's Hospital, and Speech Department, University of Miami, Coral Gables, Florida.*—Electronic measurement of nasality, desirable in clinical and medical services dealing with the speech defective, has been shown possible using the voice systems nasality meter. Previous studies indicate that the circuit, measuring phase shift of the second formant, has the potential of recognizing nasal components of the human vocal utterance. The authors developed a clinically useful test in a preliminary study. Using this on 200 normal-speaking college age adults, this report presents norms on this population. The meter measures phase shift two ways, either comparing amounts of energy stored in two separate circuits or by balancing the two circuits using a calibration dial. By the first method, males showed a mean difference of 1.2, a range from −1.0 to 3.1. Females had an average difference of 1.4, a range from −4.0 to 0.4. Using the calibration meter (reading from 0.0 to 99.9), males scored an average of 41.9, a range of 16.0 to 75.0, a standard deviation of 12.5 Females scored an average of 16.7, a range of 1.0 to 33.0, a standard deviation of 6.9. This difference was significant at the 0.002 level of confidence. Possible reasons for this difference are suggested.

FRIDAY, 17 NOVEMBER 1967 MEDALLION EAST, 9:00 A.M.

Session HH. Underwater Acoustics VI: Coherence, Fluctuations, and Marine Bio-Acoustics

HARB S. HAYRE, *Chairman*

Contributed Papers (10 minutes)

HH1. Sound Propagation Coherence Studies by Analysis of the Distortion of Explosive Pulses. R. LAVAL (nonmember), *SACLANT ASW Research Center, La Spezia, Italy.*—As an acoustic communication channel the performance of the oceanic medium is severely limited by phenomena of a random nature in its volume (temperature microstructure, interval waves, turbulences, and scatterers) and at the boundaries (surface waves and bottom roughness). In the present era of sophisticated signal-processing techniques, these coherence-destroying factors are a critical limitation, and their better understanding is necessary in order to make optimum use of the medium. An experimental method of attacking this problem is described which consists of studying the random distortion of the impulse produced by an explosive charge after propagation along the various acoustic paths. The variation in space of this distortion is considered by using long lines of hydrophones towed from the receiving ship, or vertically suspended. It is shown that most of the coherence problems can be solved by this method provided that relatively advanced techniques are available for the statistical analysis of the collected acoustic signals, and that the necessary theoretical work is carried out for the interpretation of the results.

HH2. Coherence of Convergence Zone Sound. G. R. LUND AND R. J. URICK, *Naval Ordnance Laboratory, Silver Spring, Maryland 20910*.—The vertical coherence of underwater sound in a convergence zone has been investigated by means of a vertical receiving array of six hydrophones spaced $2\frac{1}{2}$ ft apart and hung at a depth of 3800 ft from a surface ship. A second ship towed a 1120-Hz cw source outward in range from 28 to 43 miles. The outputs of the different hydrophones, when correlated, yield correlation coefficients equal to nearly unity when in the convergence half-zones at 30 and 38 miles, and show only a slow falloff with increasing separation. Between zones, where bottom-reflected paths occur, the correlation coefficient was much smaller and was found to fall off faster with increasing separation. It may therefore be concluded that a vertical array located in the convergence zone of a distant source will experience a higher signal-to-noise ratio as a result of two effects: an increased signal level due to convergence, amounting to 10 dB in our experiment, and an increased array gain due to the greater coherence of the received sound.

HH3. Measurement of Coherence in Seawater Utilizing an Acoustic Analog of Young's Experiment. J. V. LEE AND D. H. BROWN, *U. S. Navy Mine Defense Laboratory, Panama City, Florida*.—Acoustic coherence experiments have been performed in St. Andrews Bay, Florida. Phase and amplitude stability was measured in the far field of a highly directive barium titanate transducer array. Phase changes as small as 0.05° could be detected along a 7-m wavefront. The measurements were made at 200 kc/sec, depths to 7 m, and over a range of 35 m. Under typical operating conditions, the phase relation ships in the direct path and bottom bounce wavefronts were observed to be extremely stable. The results of these measurements are directly applicable in areas of interest such as: acoustic holography, the design of large-aperture synthetic or physical arrays, ultrasonic imaging, and wave propagation in random or turbulent media.

HH4. Computer Movies of Wavefront Motion. M. M. SONDHI (nonmember), *Bell Telephone Laboratories, Incorporated, Murray Hill, New Jersey*.—The motion of a wavefront in deep water can be quite complicated, owing to the variation of sound velocity with depth. To help visualize this motion, a digital computer was programmed to produce a movie of the wavefront motion for arbitrarily specified velocity variation and source position. Examples of such computer-generated movies for various velocity profiles are shown to illustrate the utility of the technique.

HH5. Fluctuation in Underwater Acoustic Propagation Over Short Depth Increments. DAVID C. WHITMARSH AND WILLIAM J. LEISS (nonmember), *Ordnance Research Laboratory, The Pennsylvania State University, University Park, Pennsylvania*—Underwater-acoustic-propagation measurements have been made off Key West, Florida periodically for the past 2 yr. Two or three deep-submergence vehicles were allowed to sink freely through the water to depths as great as 5000 ft while transponding acoustically with each other. Pulses at frequencies of 10, 20, and 40 kHz and ranges of 1700 to 4000 yd were used in the experiment. Because direct acoustic signals showed amplitude fluctuations of the order of 20 dB for depth changes 20–30 ft, a method of rapid pulsing was installed so that discrete signals 0.25 sec apart could be transmitted periodically for a period of 5 sec. This allowed a measure of the fluctuation for depth changes of only a few inches. Pulse-to-pulse amplitude differences of up to 10 dB were found for depth changes of as little as 10 in. These fluctuations are discussed in terms of transmission loss and are compared to the losses suffered for larger depth changes and those predicted by ray theory on the basis of the measured sound velocity structure of the water. [This research was supported by the U. S. Naval Ordnance Systems Command.]

HH6. Under-Ice Acoustic Stability: Results of a Preliminary Experiment and a Proposed Method of Measurement. A. R. MILNE, *Pacific Naval Laboratory of the Defence Research Board of Canada, Victoria, British Columbia*.—Sound-paths under sea-ice, in particular situations, appear to be highly stable. The consequences of high stability can be an improved target resolution in reverberation limited echo-ranging applications. A preliminary experiment to test the stability of direct sound paths under springtime shore-fast sea-ice has shown that phase shifts at a frequency of 4 kHz over $\frac{1}{2}$ mile paths were unresolved from instrumentation drift (±3°) in 8-min periods. An improved method for stability measurement is proposed. A set of N range-gated reverberation signals can exhibit "coherent" and "incoherent" components in its energy spectrum. The proportions of coherent and incoherent energy for narrow-band signals is shown to be simply related to the expectation of the time jitter of a sequence of echoes from a single reverberator located within the range gate.

HH7. Environmentally Related Amplitude Fluctuations at 420-Hz—Straits of Flordia Results. JOHN G. CLARK AND JAMES R. YARNALL (nonmember), *University of Miami, Institute of Marine Sciences, Miami, Florida* 33149.—As part of the Straits of Florida underwater sound propagation studies (Project MIMI), a 5-mo period of nearly continuous cw transmission was completed in January 1967. Since a fixed system was used for the test, the observed transmission fluctuations are solely a consequence of time-varying processes in the ocean. Statistical descriptions are being attempted of the level fluctuations at hydrophones 3 and 43 nautical miles from the source. Completed studies have defined the contribution of surface waves and internal waves to the spectrum of acoustic amplitude fluctuations. Additional analyses are expected to describe contributions at tidal and tidal harmonic frequencies. Perhaps the most significant observations are "long-term" changes in the mean propagation loss of as great as 50 dB. Available evidence points to wind associated environmental changes which can be expected to occur in many coastal areas.

HH8. Sound-Speed Measurements aboard French Bathyscaph ARCHIMEDE in the Japan Trench. K. V. MACKENZIE, *Naval Underwater Warfare Center, San Diego, California* 92152.—Data were collected to depths of 7500 m during two dives in July 1967. Sound speeds were measured with two NUS TR-4 velocimeters. Corresponding temperatures were determined with a Hewlett–Packard oceanographic sensor, a Ramsay Engineering MK VIA deep sea probe, and reversing thermometers. Pressure was determined with a Fairchild strain gauge, reversing thermometers, and a precision pressure gauge. Iron shot fouled the water-sampling system; however, data on salinity are available from previous dives. Profiles were obtained in portions of the water column and measurements of all variables were conducted at the bottom. The data were recorded as frequency on a seven-channel tape recorder during the descent and at the bottom. A reference crystal frequency was simultaneously recorded for control. The frequencies were also counted while on the bottom.

HH9. Analysis of Short Pulse Echoes from Copper Plates. F. L. WEISSER, K. J. DIERCKS, *Defense Research Laboratory, The University of Texas, Austin, Texas 78712,* AND W. E. EVANS, *Naval Weapons Center, China Lake, California 93555.*—This study was an attempt to simulate the echo-locating signal of an Atlantic bottlenose dolphin, *Tursiops truncatus,* to generate echoes from targets known to be acoustically discriminable by an animal of this species. The targets were flat copper plates that varied in the thickness dimension only. An 0.22-cm thick plate was selected as reference; the animal was unable to discriminate an 0.16-cm-thick plate from the reference plate, but was able to discriminate 0.28 and 0.32-cm thick plates on 60% and 75% of the trials, respectively (Evans and Powell, "Discrimination of Different Metallic Plates by an Echolocating Delphinid," *Animal Sonar Systems, Proceedings of the Symposium on Bionic Models of the Animal Sonar System,* Frascati, Italy, 26 Sept.–3 Oct. 1966). Echo waveforms and scattering patterns were recorded at 6-in. range increments from 14 in.–50 in. test ranges. Pulse repetition rates and signal levels similar to those of the animal were used. The scattered field was sampled using biaural hydrophones separated approximately the distance of the animal's receptors. Only monaural signal characteristics are reported here. Results of the analysis show power spectral density to be the most promising discriminant (target-specific) characteristic. Differences in scattered intensity are noted, but are slight. No obvious differences between echo waveforms from the different plates were observed. Correlations with the outgoing signal display discriminable peak values, which may be artifactual. [This work was supported by the Office of Naval Research.]

HH10. 20-Hz Signals Observed in the Central Pacific. JOHN NORTHROP, *University of California, San Diego, Marine Physical Laboratory of the Scripps Institution of Oceanography, San Diego, California 92152,* W. C. CUMMINGS, AND P. O. THOMPSON, *U. S. Navy Electronics Laboratory, San Diego, California 92152.*—Twenty hertz signals, similar to those reported by other investigators from different locations, were recorded on hydrophones of the Pacific Missile Range at Midway, Wake, and Oahu Islands. Year-round recordings revealed the signals during the winter months with a peak in October at Midway, December at Oahu, and January at Wake. There was no daily pattern of occurrence. More signals were recorded at Midway than at the other locations. Signals appeared in pairs with members separated by 16 sec between Type I and Type II, and 20 sec between Type II and Type I. Both signals had maximum energy near 20 Hz and lasted about 1 sec. Type I ranged from 18–27 Hz; whereas Type II ranged from 18–44 Hz. Signal trains lasted 10–12 min, and they were separated by 1–3 min. Estimated source levels ranged from 85–120 dB *re* 1 μ (at 1 yd). Source movement was evident at stations where a series of signals was recorded at two or more hydrophones.

HH11. Sound Production of Migrating Gray Whales, Eschrichtius gibbosus Erxleben. W. C. CUMMINGS, P. O. THOMPSON, AND R. D. COOK (nonmember), *U. S. Navy Electronics Laboratory, San Diego, California 92152.*—Observations were made during two southward breeding migrations (1966, 1967) of gray whales. Bottom-mounted hydrophones and a moored ship were positioned off San Diego, in the path of about 230 whales. Sound sources were located using received levels and arrival time differences. Visual tracking corroborated sound data. *Moans* were, by far, most common of more than 200 whale sounds. *Moans* lasted 1.25 sec; their source level was about 120 dB *re* 0.0002 dyn/cm² at 1 yd; and they ranged 20–150 Hz. Whales moaned while submerged, day and night, under varied weather conditions. *Blows* (surface exhalations) lasted 1.5 sec and ranged 15–175 Hz. Infrequent *bubble-type signals,* lasting 0.7 sec, were about 112 dB *re* 0.0002 dyn/cm² at 1 yd, and ranged 15–305 Hz. Average swimming speed of single whales was 5.5 knots, based on sound tracks. Whales migrated all night despite dense fog. Large bioluminescent areas around migrating whales were seen from the ship and from an airplane. No characteristic behavior could be associated with sound production.

HH12. Sound Fields within Aquaria. ANTARES PARVULESCU. *Hudson Laboratories of Columbia University, Dobbs Ferry, New York, 10522.*—Small aquaria are generally convenient for bioacoustic experiments. Sound introduced into such tanks is often measured with pressure-sensitive hydrophones. Unfortunately, sound in a small tank never propagates in a plane wave, and therefore pressure measurements can be highly misleading. The sound field also fails to resemble the natural acoustic environemnt of the organisms. We have obtained further measurements of the effective wall impedance and, hence, estimates of the wave impedance within such tanks, using the technique described at the Second Symposium on Marine Bio-Acoustics (1966). The fields within the tanks are similar whether the sound source is located in the water or in the air nearby. Particle velocities in the water (and hence displacements) are orders of magnitude greater than has been sometimes assumed from uncritical use of pressure hydrophones. This suggests that animal behavior may be due to lateral-line response rather than to auditory system response. [Hudson Lab. of Columbia Univ. Informal Documentation No. 148. This work was supported by the U. S. Office of Naval Research.]

HH13. A Test Tank for Independent Determination of Pressure and Velocity Thresholds of Small Fish. WILLIAM SILER (nonmember), *Downstate Medical Center, Brooklyn, New York,* PHYLLIS CAHN (nonmember), *Yeshiva University, New York,* JEROME WODINSKY (nonmember), *Brandeis University, Waltham, Massachusetts,* AND JAMES FITZGERALD, *Fitzgerald Laboratories, Annapolis, Maryland.*—A test tank is described which permits individual variation of pressure and velocity sound-wave components. Standing waves are generated by two J9 speakers with precisely controllable relative amplitude and phase. A velocity unit analogous to the microbar is proposed to be used for calibration, and a Bode-type plot is presented for relating test tank conditions to the spatial near or farfield of traveling waves. Use of the tank in fish threshold determination is described. Analysis of threshold test data is complicated by the "two-stimulus" problem, and a graphical analysis method is presented. Strengths and weaknesses of the prototype tank are discussed, and plans for approaching the general noise problem discussed. [Work supported by the National Science Foundation.]

FRIDAY, 17 NOVEMBER 1967 EMPIRE ROOM, 9:00 A.M.

Session II. Engineering Acoustics IV: Transducers and Materials

CLAUDE C. SIMS, *Chairman*

Contributed Papers (10 minutes)

II1. Nonlinear Distortion in Dynamic, Direct Radiator, Loudspeakers. HARRY F. OLSON, *RCA Laboratories, Princeton, New Jersey.*—The major contributors to nonlinear distortion in dynamic direct radiator loudspeakers are the driving and suspension elements. These elements are constant for small and moderate amplitudes but depart from constancy for large excursions of the cone and voice coil. A theoretical analysis shows that the major spurious components due to the nonlinear distortion are third harmonics of the fundamental. Means for reducing and canceling the nonlinear distortion due to the driving and suspension systems are described.

II2. An Infrasonic Generator. W. C. RICHIE (nonmember) AND C. P. LITTLE (nonmember), *Department of Engineering Science, Trinity University, San Antonio, Texas* AND D. W. EVERTSON, *Defense Research Laboratory, University of Texas, Austin, Texas.*—An infrasonic generator utilizing a single bellows driven by a linear displacement transducer has been constructed. The generator has a dynamic range that extends from 0.5 dyn/cm^9 to 600 dyn/cm^2 and a frequency response of +0.5 dB, −0.6 dB from 0.001 Hz to 10 Hz. The infrasound generated by this device will be utilized for the study of the effects of infrasound on human beings. [This research is supported by National Science Foundation Grant 6K 1588].

II3. Investigation of the Vibrational Properties of a J9 Transducer in Water by an Optical Interferometric Technique. J. J. GILHEANY (nonmember), AND F. A. ANDREWS, *The Catholic University of America, Washington, D. C. 20017.*—The magnitude and phase of the mechanical and radiation impedance of a USRL J9 transducer at nonresonant frequencies have been directly measured under free-field conditions by an optical interferometric technique. The requisite facility and technique (measurement of vibrational amplitude as a function of drive current) are described. The mode shape of the transducer in air and in water is also shown. [Work supported by Code 468 of the Office of Naval Research, U. S. Navy.]

II4. Transducer Element Design for an Endfire Array. HARRY B. MILLER, *Electronics Division General Dynamics, Rochester, New York.*—In the design of resonant transducer elements for a high-power array steerable from broadside to endfire, some new design concepts and terminology have arisen such as velocity control, Z_{oc}, etc. The new approach tends to avoid *explicit* use of such design parameters as coupling coefficient, mechanical Q, etc. [H. B. Miller, High Power Steerable Planar Array, JUA(USN) 14, 401 (April 1964)]. The question arises as to whether the new approach alone is sufficient for the designer.

II5. Design of a High Amplitude Resonator. J. G. MARTNER (nonmember) AND S. V. HANAGUD (nonmember), *Stanford Research Institute, Menlo Park, California 94025.*—Design parameters and operation of a new type ultrasonic resonator are described. The resonator consists of a solid cylindrical body that contains a centrosimetric conic cavity, and is driven by a piezoelectric disk bonded to the base of the body. The system thus generated vibrates in a radial mode and the maximum amplitude is measured at the rim of the system. Optimum vibration conditions are obtained when the ratio of height to radius of the resonator is 1.24. An equation is derived for the frequency of vibration as a function of the properties of the material and the physical dimensions. A number of models were constructed and their capabilities as atomizers were studied. Aerosol conversion capabilities were found to be in excess of 22.5 gal of waterlike liquid per hour with an aerosol-generating area of approximately 45 cm^2. The design of fourth enharmonic resonators whose cross sections are similar to the ones described is briefly discussed.

II6. A Finite Difference Analysis of the Vibrations of Solid Cylinders. G. W. MCHAHON, *Naval Research Establishment of the Defence Research Board, Dartmouth, N. S., Canada.*—A finite difference analysis has been used to study the free axially symmetric vibrations of solid isotropic elastic cylinders. A grid having up to 36 independent lattice points was used with simple central difference formulas to set up a system of homogeneous equations, whose determinant was solved to yield the resonance frequencies. The results have been compared with Rayleigh-type thickness corrections to the frequency equations for thin disks and slender rods, and with experimental observations on solid steel and aluminum cylinders. The results indicate an accuracy within 1% for the first longitudinally symmetric mode and within 4% for the second-symmetric and first- and second-antisymmetric modes over a height-to-radius ratio range of 0.8 to 3.5 and for Poisson's ratio up to 0.34.

II7. Mechanical Stability of the Diaphragm of an Electrostatic Loudspeaker. JOSEF MERHAUT, *Technical University, Prague, Czechoslovakia.*—Mechanical stability of thin diaphragm in an electrostatic loudspeaker depends on the balance between the mechanical tension of the diaphragm and the electrostatic forces between it and the back electrodes. The condition of this balance as a function of the design parameters is calculated. The author shows in a push–pull electrostatic loudspeaker, that the requirements for mechanical stability and the lowest-frequency limit are contradictional. Futher, the formula for this frequency limit in connection with the condition of mechanical stability of the diaphragm is derived.

II8. Dynamic Mechano-Electrical Cable Noise. J. E. DONOVAN, *Underwater Sound Reference Division, Naval Research Laboratory, Orlando, Florida 32806.*—Measurements have been made on 11 samples of commercially available electroacoustic transducer cables to determine their electrical output as a function of dynamic mechanical deformation (i.e., bending twisting, crushing). Included were two " low-noise" cables. Voltages generated ranged from less than 10 μV to several hundred mV peak to peak when the cables were mechanically driven at 15 Hz. The outputs as a function of time varied over several orders of magnitude. This is an amplification of the work done by Perls [J. Appl. Phys. 674–680 (June 1952)], Rasmussen [Bell Lab. Record 305 (Aug. 1959)], and Hole [Elec. Eng. 770 (Dec. 1960)], with emphasis on the comparative effects of controlled parameters, especially time.

II9. Transducer Pressure-Release Material for Deep Submergence Applications. SHELBY F. SULLIVAN (nonmember) AND HARPER JOHN WHITEHOUSE, *Naval Undersea Warfare Center, Pasadena, California 91107.*—A new acoustic pressure-release material is proposed for use as a reflective backing in deep submergence transducer arrays. The material is high-temperature aviation insulation manufactured by Johns–

Manville under the trade name Min-K 2000, available in several densities between 20 lb/ft³ and 35 lb/ft³. As purchased, the material is inelastic and is prestressed for transducer applications by uniaxial compression to a pressure greater than the design depth of the array structure. The material takes a permanent set depending on the initial density, the resulting material having a density that depends on the inelastic prestress and not on the initial density of the parent material. Similarly, acoustic velocity for longitudinal waves is dependent only on the inelastic prestress. The pressed material is rigid and machinable, having about half the impedance of acoustic backing paper at the same operating depth. A test array fabricated with this material and containing an air-filled cylinder at its center showed a 20 $\log_{10} E/E_0$ minor lobe level of -27 dB (the design level was -30 dB). The array was essentially pressure insensitive to the 2000-psi limit of the transducer test facility.

II10. Measurement of the High Pressure Acoustic Properties of Various Transducer Materials. R. W. HIGGS, *Honeywell Corporate Research Center, Hopkins, Minnesota.*—An impedance tube technique is used to measure the acoustic propagation constants and characteristic impedance of various transducer materials as a function of hydrostatic stress and frequency. The materials measured function as reflectors, decouplers, absorbers and windows in the transducer assembly and the high-pressure properties are required to assure high operational efficiency and satisfactory directivity patterns in deep ocean operation. The pressure in these measurements is varied from 10 to 45 kHz. The acoustic measuring system is described and the characteristics of the materials are presented.

II11. Some Acoustic Source Experiments Conducted in 1966. WILLIAM H. CRAFT, *Lockheed Missiles and Space Company, Huntsville, Alabama.*—In recognition of the need for a better high-intensity acoustic source capable of simulating rocket engine noise and in-flight aerodynamic noise, Lockheed conducted an experimental program during 1966 in an attempt to develop such a source. Most of the experiments performed were of a "Combustion Modulation" principle whereby acoustic energy is produced through control of the reactants in a combustion process. Peripheral tests were made including attempts to modulate low-volume air flow. In one peripheral experiment, a discovery was made that a worthwhile acoustic source is achieved when a stream of hydrogen and oxygen in detonatable proportions is subjected to a repeating electrical spark. Sonic pulses are produced at the frequency of the spark repetition rate. The device, called the "sonic detonator," promises early application as a source of high-intensity, high-energy acoustic pressure. The paper describes the tests performed and results achieved.

II12. A Family of Electrostatic Sensors for Hypervelocity Microparticle Impact Momentum Measurement, N. J. MEYER, B. R. BEAVERS, AND J. G. POWELL, *LTV Research Center, Western Division, Ling-Temco-Vought, Inc., Anaheim, California.*—The basic transduction principles of the condenser microphone have been adapted to produce a suite of hypervelocity microparticle impact sensors. Two types of sensors have evolved from the basic condenser microphone geometry. *First*, a lumped parameter (rigid impact plate supported by a spring) device analogous to the ballistic pendulum, and *second*, a distributed parameter unit of maximum sensitivity that senses the capacitance variations produced by the particle impacts on an extremely thin plate. In the case of the lumped parameter design, laboratory tests have verified the theoretically predicted sensitivity. Variations on the basic lumped parameter design have permitted momentum measurements over a range of seven orders of magnitude, while the distributed parameter model has demonstrated a sensitivity at least two orders of magnitude better than any device yet reported. [This work, which has been partially supported by NASA Manned Spacecraft Center, is preparatory to the measurement of particle impacts on the lunar surface.]

II13. Small, Low-Frequency Omnidirectional, Broad-Band Hydrophone. RUELL F. SOLBERG, JR. (nonmember),* *Defense Research Laboratory, The University of Texas at Austin,* AND LEONARDT F. KREISLE (nonmember), *Department of Mechanical Engineering, The University of Texas at Austin, Austin, Texas 78712.*—A small, low-frequency omnidirectional, broadband hydrophone with relatively good sensitivity and capacitance has been designed, constructed, and tested. The design was of such nature that the hydrophone should withstand vibration and high-velocity shock in addition to meeting other particular requirements. A radially polarized, PZT-5H piezoelectric ceramic tube with end caps and an internal air cavity was cast in a polyurethane compound. This piezoelectric material has a high permitivity for maximum capacitance for permitted physical dimensions; however, two disadvantages are its low upper Curie point and its large variation of capacitance with temperature. A 1-in. outside diam, $\frac{1}{8}$-in. wall thickness, and 1-in. length tube was used as the active part of the hydrophone. Properties of the hydrophone include a capacitance of 0.013 μF, sensitivity of about -95 dB, and omnidirectionality in all planes to about 5 kHz. [The work was partially supported by U. S. Naval Ordnance Laboratory.]

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FRIDAY, 17 NOVEMBER 1967 — SILVER CHIMES EAST, 2:00 P.M.

Session JJ. Noise IV: Aircraft Acoustics and Aerosonics

HARVEY HUBBARD, *Chairman*

Contributed Papers (10 minutes)

JJ1. Penetration of a Sonic Boom into Water, by RICHARD K. COOK, *Environmental Science Services Administration, Rockville, Maryland, 20852.*—An aircraft in level flight at a supersonic speed over water produces a sonic-boom pressure on the water surface. Usually the N wave of the incident sound pressure moves across the surface at a subsonic speed, relative to the speed of sound in water. Therefore, the boom energy is totally reflected, and the sound pressure in the water falls off exponentially with depth below the surface. The penetration depth, comprising about 85% of the underwater energy, is of the same order as the N-waveform length on the surface. The analysis shows further that the underwater N wave has a weak precursor and a weak tail. The same is true of the wave reflected into the atmosphere. The N-wave pressure is an odd function of distance from the wave center, but the precursor and tail is an even function. This is a special case of reflection effects on a sonic boom. If small and brief (in time) perturbations occur at a reflecting surface, the perturbation pressures will be approximately the same function of distance from the trailing edge of the boom as they are from the leading edge.

JJ2. Factors Contributing to the Loudness of Sonic Booms. P. B. ONCLEY, *The Boeing Company, Seattle, Washington.*—Most of the flyover tests on subjective reaction to sonic booms have relied only on peak overpressure as a measurement standard. Loudness is, however, greatly influenced by wave shape, and by rise time in particular. Oncley and Dunn [J. Acoust. Soc. Am. (to be published)] have derived a simple expression for the energy spectral density of N waves with finite rise time. Loudness computations based on this expression agree well with the experimental results of Zepler and Harel, and also explain the extreme variation in loudness within a comparatively small range of overpressure. Actual sonic-boom analyses frequently show rise times of 10 msec or more, generally increasing with altitude. Since flyover tests have usually used fighter aircraft at 30 000 to 40 000 ft to simulate the pressure levels at twice that altitute from the SST, it is probable that on the average, booms were louder and sharper than the SST would produce. Conditions under which boom loudness can be reduced by configuration control will be considered.

JJ3. Total Reflection of the Sonic Boom. C. I. BEITIN (nonmember), *Bell Telephone Laboratories, Inc., Whippany, New Jersey.*—A method based on the linear theory of acoustics has been developed to determine changes in sonic boom signatures as a result of total reflection at the earth's surface. It can be shown that for Mach reflection when the slope of the terrain deviates less than 3°, from the horizontal, or for nearly grazing incidence, large spikes appear at the leading and trailing fronts of the signature. The size of these spikes, may be as high as five times the amplitude of the incident wave. The results agree very well with field measurements.

JJ4. An Experiment to Locate The Acoustic Sources in a High Speed Jet Exhaust Stream. R. C. POTTER, *Wyle Laboratories Research Staff, El Segundo, California 90246* AND J. H. JONES, *NASA, Marshall Space Flight Center, R-AERO-AUA, Huntsville, Alabama 35812.*—A small high-speed (Mach 2.5) cold jet was operated with the exhaust stream passing through a hole in the wall of a 100 000 ft reverberation room. The reverberant SPL was measured to allow the total acoustic power generated by the flow inside the room to be determined. The jet nozzle was then progressively withdrawn so that a smaller amount of the mixing flow was retained within the room. The experiment was repeated with the jet inside the room, with the flow directed outwards and the jet nozzle moved back into the room. The resultant acoustic power curves were differentiated to give the acoustic power generated per unit length of the jet gas flow. The results are presented and experimental details reviewed. It is concluded that the flow near the supersonic core tip is responsible for the majority of the noise generated.

JJ5. Wind Tunnel Study of Probes for Measuring Pressure Fluctuations on an Ablative Hypersonic Reentry Vehicle. H. H. HELLER, *Bolt Beranek and Newman Inc., Cambridge, Massachusetts 02138.*—The excessive heat development on the exterior structure of hypersonic reentry vehicles necessitates the use of heat-resistant pressure transducers for the measurement of boundary layer pressure fluctuations on the vehicle's surface. An appropriate acoustic probe configuration was developed and tested in windtunnel experiments. The frequency response of the probe configuration to flow excitation was studied under subsonic ($M = 0.055$) and supersonic ($M = 2.92$) flow conditions and was found to depend both on the probe interior geometry and on the physical characteristics of the flow. In addition, the influence of heat-shield ablation on the frequency response was studied under these same conditions. Widnall and Lyon's theory of the dynamics of the probe in the presence of hypersonic flow was confirmed by the experimental findings. [This work was sponsored by the Space and Missile System Organization.]

JJ6. The Effect of Exhaust Temperature on Jet Noise Generation. H. E. PLUMBLEE, JR., *Aerospace Sciences Laboratory, Lockheed-Georgia Company, Marietta, Georgia.*—An experimental study was made to determine the effects of jet engine exhaust temperature on jet noise generation. Noise data were taken at 70 points on a uniform grid in the nearfield and farfield of a 3.5 in. diam model jet. Jet efflux temperature was varied from 560°R to 3860°R and exhaust velocity ranged from 930 fps to 4660 fps. Based on existing theories, a semiempirical relationship was derived for predicting the nearfield and farfield sound pressure levels. A computer program was subsequently developed which calculates and plots SPL contours for the over-all and octave bands, using exhaust temperature and Mach number as controlling parameters. In the farfield, a unique method is shown for reducing jet noise. The reduction is based on fixed exhaust velocity and thrust while varying temperature. A reduction of 5.5 dB has been noted for a 130% increase in jet efflux temperature, i. e., 560°R to 1290°R. This is a continuing research program and results of tests designed to further define the reduction phenomenon will be included, if available. [Portions of this research were sponsored by Air Force Contract AF33 (615)-2503.]

JJ7. Noise Measurements During Take-Off and Climbout Operations of Jet Transports. W. LATHAM COPELAND, *NASA-Langley Research Center, Hampton, Virginia.*—Data were obtained under closely controlled conditions for both simulated and actual climbouts with the objective of correlating the operation of the aircraft with the measured noise. Two-, three- and four-engine commercial jet aircraft having turbojet and turbofan engines were included in the program. Good correlation was found between the ground noise measurements and such factors as flight profile, engine power, flap setting and aircraft speed. Parametric charts are included to illustrate the relative significance of the above variables and are shown to be useful for ground noise predictions.

JJ8. Influence of Airspeed on Noise Generated Within Various Fixed- and Rotary-Wing Aircraft. DONALD C. GASAWAY, *USAF School of Aerospace Medicine, Brooks Air Force Base, Texas.*—The data reported in this study will assist human-factors engineers and aeromedical personnel in assessing noise exposures associated with the operation of aerospace vehicles. The influence of increased flight speeds on the level and character of noise generated within the cockpits of 148 fixed- and rotary-wing aircraft is described and illustrated. Stereotyped collateral effects resulting from increases in airspeed are categorized for different types of vehicles and the noise envelopes that evolved from this study are grouped into two main categories: fixed-wing and rotary-wing. Secondary noise envelopes are then illustrated for each of the two main groups according to the type of power plant mated to the vehicle. The envelopes derived from this study clearly illustrate differences in the character and level of noise generated within various aircraft due to increased airspeed. Also, and probably more important, the various envelopes verify that within similar types of aircraft the increased airspeed results in changes in noise levels and spectra that are stereotyped.

JJ9. Interior Noise of Airline Aircraft. RICHARD B. STONE (nonmember), *Air Line Pilots Association*.—Data gathered with an octave band analyzer is reported for many of the currently operated propeller, turboprop, and jet airline aircraft. The investigation dealt primarily with noise associated with the cockpit environment, however, a number of noise profiles in the passenger cabin are presented. Two criteria are the basis for evaluating the environment: damage risk criteria and speech interference level. Almost all airline cockpits rely on the unaided voice to communicate commands and therefore interference with speech is critical. Even recent models of jet aircraft cause cockpit communications, in some regimes of flight, to be carried out at a near shout. Many of the relatively new turboprop aircraft exceed the damage risk criteria in the cockpit.

JJ10. On the Acceptability of Aircraft Noise Heard in the Presence of Speech. CARL E. WILLIAMS, KENNETH N. STEVENS, AND MARY M. KLATT (nonmember), *Bolt Beranek and Newman Inc., Cambridge, Massachusetts 02138*.—Various aircraft flyovers were presented to listeners who were asked to rate them in terms of their acceptability in the home. Ratings were assigned on a scale having four categories: "of no concern," "acceptable," "barely acceptable," and "unacceptable." Judgments were obtained for two listening situations: (1) speech presented simultaneously with the noise, and (2) noise presented without speech. For a given maximum noise level, little difference was observed between ratings obtained in the absence of speech and ratings obtained with speech present at a comfortable level. When the noise reached a level that caused comprehension of the speech to deteriorate, it was judged to be worse than "barely acceptable." Other data indicate that such a noise level produces a sharp drop in sentence intelligibility. Speech interference, whether actually present or estimated on the basis of past experience, appears to play a role in shaping the judgments individuals make regarding the acceptability of aircraft noise heard indoors. [K. N. Stevens is also at the Massachusetts Institute of Technology. Work supported by the Federal Aviation Agency.]

JJ11. Laboratory Study of Jet Aircraft Noise Complaint Potential as Related to Context of the Noise. J. E. MABRY, J. W. LITTLE (nonmember), *The Boeing Company, Seattle, Washington*.—Previous studies using category scaling of jet aircraft flyby noise have emphasized scales of annoyance, objectionability, noisiness, intrusiveness, and so on. The present study related subjects' predictions of their complaint performance to acoustical characteristics of recorded flybys. In addition to the aim of obtaining a direct relationship between "expectation of making a complaint" and acoustical parameters, the study investigated predicted complaint performance as related to the immediate acoustical environment. For example, there were significant differences in complaint predictions for the same flyover when juxtaposed (both before and after) with different flyovers; also, the judgment that a particular flyover was more annoying than a comparison flyover increased significantly the probability of predicting a complaint for that flyover. This is laboratory data that nicely supports the "projection hypothesis" of the 1961 Farrborough experiments. Subjects' complaint predictions from the laboratory setting are also related to actual complaint performance obtained from sociological survey type investigations. Contrary to the findings of previous investigations, complaint predictions for males were significantly greater than for females.

JJ12. Evaluation of Fifteen Scaling Units for Estimating the "Annoyance" of Jet Aircraft Flyovers. J. W. LITTLE (nonmember), J. E. MABRY, *The Boeing Company, Seattle, Washington*.—Using the psychophysical method of constant stimuli, 36 subjects (18 males and 18 females) under two sets of instructions, made annoyance judgments for a total of 8496 pairs of recorded actual flyovers. Flyovers were selected on the basis of magnitude of tone element and 20 dB down duration. Fifteen different scaling units purporting to measure the relationship between physical acoustical properties and "annoyance" were applied to the various flyovers. Some of the scaling units applied were PNdB, PNdB with a tone correction, Effective PNL, Composite PNL, Stevens' phons, Stevens' phons with a pure tone correction, and so on. After applying each of the 15 different scaling approaches to the flyovers, each approach was related to the annoyance judgments obtained. Surprisingly, Stevens' phons with a pure tone correction predicted equal annoyance points better than any of the other 14 approaches. Possible application of the results to noise suppression devices for jet engines is also discussed.

FRIDAY, 17 November 1967 MEDALLION EAST, 2:00 P.M.

Session KK. Psychological and Physiological Acoustics VIII: Frequency and Pitch

JOHN BRANDT, *Chairman*

Contributed Papers (12 minutes)

KK1. Pitch of Pure Tones. ELCA SWIGART (nonmember), *The Ohio State University, Columbus, Ohio*.—In Part I, university students enrolled as music majors served as subjects to determine discrepancies between frequencies of pure tones presented through either a bone oscillator or earphones (monaurally and binaurally) and the fundamental frequency of vocal limitations of pitches of the pure tones. All vocalizations were recorded on magnetic tape and analyzed to determine fundamental frequency. In Part II, pure tones were presented at one of two intensities through a bone oscillator placed on the mastoid bone and matched in pitch with a pure tone (variable in frequency by the subject) presented through earphones (monaurally and binaurally). Discrepancies of responses in hertz from the stimulus were recorded by the examiner. Results indicated minimal differences among responses to stimuli through the bone oscillator and monaurally and binaurally through earphones. However, an increase in intensity of the stimuli through the bone oscillator raised the matched frequency of pure tones through the earphones presented either monaurally or binaurally. An increase in intensity of stimuli presented binaurally through the earphones lowered the fundamental frequency of vocal limitations of those pitches. Analyses of pitch matching responses to other stimulus conditions were also computed.

KK2. Perceived Order of Tone Pulses. ROBERT PETERS, *University of Southern Mississippi, Hattiesburg, Mississippi.*—Order of a series of tone pulses is perceived differently from actual order when frequency of the pulses vary. If the center pulse differs in frequency from the other pulses in a train, the center pulse is heard as either first or last in the series. Because this was observed when the frequency of a center pulse was adjusted for trains, the effect may have been a function of repetition. For this study, 12 subjects judged the order of pulses in trains presented in random order. There were three pulses in each train each of 20 msec duration with pulse intervals of 50 msec. The first and third pulses were 1000 Hz in all cases; the center pulse was varied in discrete steps from 400 to 2000 Hz. The data indicated that perceived order was controlled by frequency of the center pulse. For center pulse frequencies below 600 and above 1600 Hz the observers judged that the center pulse followed the other two pulses. [This work was supported by the U. S. Air Force under Contract No. AF 33 (615)-1181.]

KK3. Perception of Frequency Modulation. YUKIO TAKEFUTA, *The Ohio State University, Columbus, Ohio.*—The purpose of this experiment was to plot equal pitch-modulation contours as statistical functions of duration, direction, and extent of frequency modulation. Twenty college students listened to 66 quasipure tones each of which was a combination of one of six durations (40–1000 msec) and one of 11 extents of modulation (-10 to $+10$ semitones). Duration and direction as well as extent of frequency modulation affected pitch modulation as evaluated by a method of magnitude estimation. A regression line was fitted to the listeners' responses plotted on a coordinate of pitch modulation (psychological) and frequency modulation (physical) for each combination of duration and direction. Expected mean values on the regression lines rather than the actual mean values were used to plot the contours for each extent, duration, and direction. The nearest but significantly different mean values from the expected mean of zero were used to determine the thresholds of pitch modulation. The distance between the threshold value and zero mean was determined by calculating confidence intervals for the expected mean values of zero by using the variance of the observed data.

KK4. Detection and Relative Discrimination of Auditory "Jitter." IRWIN POLLACK, *Mental Health Research Institute, University of Michigan, Ann Arbor, Michigan.*—Thresholds for the minimal departure from periodicity and thresholds for discrimination of relative departure from periodicity were obtained in order to explore the limits of temporal resolution for auditory pulse trains. In a forced-choice test, listeners were presented pulse trains, one of which was subjected to more variability, or jitter, than the other three. The principal experimental variables examined were: the number of intervals; the mean interpulse interval; and, the jitter of the reference pulse train. Minimum thresholds for jitter, relative to the center interpulse interval, is less than 0.1% for a large number of pulses. Threshold jitter decreases with shorter interpulse intervals for a large number of pulses. Jitter thresholds are minimal for interpulse intervals of 4–6 msec for a small number of pulses. Jitter discrimination is approximately independent of the reference jitter for a small number of pulses and is nearly directly proportional to the reference jitter for a large number of pulses. The temporal precision of the auditory system, in contrast with its precision of spectral analysis, appears insufficient to account for minimal jitter thresholds.

KK5. Frequency Discrimination as a Function of Frequency.* G. BRUCE HENNING AND S. M. FORBES (nonmember), *Defence Research Establishment Toronto, Downsview, Ontario.*—The frequency resolution of the human auditory system at high frequencies has been difficult to measure because of the nonlinear response of earphones on real ears at frequencies above approximately 4000 cps. Variation of this response with frequency is sufficiently great that amplitude rather than frequency differences may govern discrimination. An experiment with random amplitude signals [Henning, J. Acoust. Soc. Am. 39, 336–339 (1966)] indicated that frequency discrimination at high frequencies is almost an order of magnitude poorer than previous measures indicate. This finding is confirmed in experiments in which the amplitudes of the signals of different frequency to be discriminated are equalized under the earphones by means of a small probe microphone.

* Defence Research Establishment Toronto, Research Paper 675.

KK6. Identifying Meaningless Tonal Complexes. J. C. WEBSTER,* ALAN CARPENTER (nonmember), AND M. M. WOODHEAD (nonmember), *Applied Psychology Research Unit, Cambridge, England.*—Series of buzz tones with up to 24 harmonics were presented to five groups of listeners for identification. Nine tones in which different harmonics were emphasized were presented to Group I who could easily tell them apart and could identify them with 33% accuracy. The remaining groups heard three of the tones in various conditions of noise, filtering, fundamental frequency, and emphasized to nonemphasized harmonic intensity differentials. The most difficult task was to identify the tones with differing harmonic structure when the fundamental frequency was not the same for every complex tone. It was also difficult to pick out complexes with the same harmonic structure and basic frequency when noise-masked, filtered, and in-quiet items were intermixed at random in the same test. The distinguishing differences in the meaningless buzz complexes were in harmonics six and above.

* Exchange scientist from U. S. Navy Electronics Laboratory, San Diego, California.

KK7. Aural Harmonics: Tone-on-Tone Masking at Various intensities of the Fundamental. T. D. CLACK, *Kresge Hearing Research Institute, University of Michigan Medical School, Ann Arbor, Michigan, 48104.*—Aural harmonic distortion was produced in seven normal ears by presenting a 1000-Hz pure tone (f_1) at 57, 61, 65, 67, 69, 71, and 74 dB SPL. Simultaneously, listeners traced a Békésy-type threshold for an objective 2000-Hz tone (f_2) which was "phase shifted" through 360°. The variations in such tracings fit the equation; $Y=A_1+A_2 \sin(x+\phi)$. The effect of the changes in f_1 intensity upon the three dependent variables, A_1, A_2, and ϕ, will be analyzed separately. Results indicate that increases in the f_1 intensity produce: (1) linear increases (slope$\approx$2.0–2.5 dB) in A_1, the over-all masking level; (2) a small but complex change in A_2, the phase amplitude effect; and (3) no change in ϕ, the phase shift. These data will be interpreted in terms of a model that has three major assumptions: (a) the ear generates an aural harmonic (AH); (b) the objective f_2 and the AH sum vectorially; (c) the listener must maintain the resultant sum at some perceptual threshold level.

FRIDAY, 17 NOVEMBER 1967 BURGUNDY EAST, 2:00 P.M.

Session LL. Engineering Acoustics V: Signal Processing, Acoustical Instruments and Measurements

G. A. SABIN, *Chairman*

Contributed Papers (10 minutes)

LL1. A New Technique for Determination of Cross-Power Spectral Density with Damped Oscillators. WAYNE E. SIMON (nonmember), *Martin Marietta Corporation, Denver, Colorado* AND LOUISE A. WALTER (nonmember), *Denver Research Institute, University of Denver, Denver, Colorado.*—A new technique for computation of cross-power spectral density of random processes has been developed, programmed, and tested on a variety of constructed random and periodic signals of known characteristics. The technique is based on derived relationships between cross-power spectral density and the mean value of certain products of a random signal and the response of an oscillator driven by the random signal. In addition, a technique for discrimination between periodic and random signals has been developed that is based on comparison of the amplitude distribution of the signal and the amplitude distribution of the acceleration of an oscillator driven by the signal. Computations have been performed for a variety of signals with known characteristics with very satisfactory results.

LL2. Bandwidth Limitations in Measurements of Cross-Spectral Density. H. COX, *Naval Ship Systems Command, Washington, D. C. 20360* AND M. STRASBERG, *Naval Ship Research and Development Center, Washington, D. C. 20007*—This paper discusses the bandwidth requirements for accurate estimates of cross-spectral density. The common way to reduce the statistical fluctuations occurring in measurements of auto- and cross-spectral density is to increase the analysis bandwidth and accept the resulting decrease in resolution. For cross-spectral estimates, however, there is a danger that the estimates will be low, low by perhaps orders of magnitude, if the analysis band is too wide. The band must be so narrow that both the phase angle and the magnitude of the cross spectrum remain relatively constant over the band. Because of the phase requirement, estimates of cross spectra usually require narrower bands than do autospectral estimates (which are insensitive to phase). This limitation is common to both digital and analog estimates. Several case histories are discussed where errors resulted from excessively wide bands. The bandwidth limitations associated with several physical systems are discussed in terms of phase velocities. It is shown that estimates of ocean-wave cross spectra require that the bandwidth *decrease* with increasing frequency. Finally, it is shown how the errors can be reduced by use of phase equalization.

LL3. In-Flight Spectrum Analysis of Random Waves. T. COFFIN AND J. SEROCKI, *Chrysler Corporation, Space Division; Huntsville, Alabama 35807.*—Direct transmission of wide-band spacecraft data places an inordinate burden on the available telemetry spectrum. It is extremely advantageous from the standpoint of telemetry spectrum economy to process measurements onboard and transmit the reduced data over a narrow-bandwidth telemetry channel. This paper describes a random wave analyzer for inflight processing of vibration and acoustic data. Development considerations are discussed and an evaluation of instrumentation concepts presented. A functional description of the technique selected for implementation is presented. The technique may be described as an *incremental constant bandwidth analyzer*, since three synchronized sweeping filters are utilized to produce a spectrum with differing resolution in separate frequency increments. Typical analysis times range from 1 to 8 sec. Sweep rates, averaging times, and frequency increments are adjustable, providing maximum flexibility. An evaluation of the analyzer is summarized and spectral estimates compared with those obtained by a laboratory quality wave analyzer. Physical characteristics of a six channel prototype system are described. [Sponsored by NASA Manned Spacecraft Center.]

LL4. A Real-Time Debye–Sears Effect Spectrum Analyzer for Audio Frequencies. F. H. SLAYMAKER, *Electronics Division of General Dynamics.*—The Debye–Sears effect is usually thought of as involving the generation of a Fraunhofer diffraction pattern (the Fourier transform) from the spatial phase variations produced in a coherent light beam that has passed through an ultrasonic beam in a direction parallel to the sound wave front. For a typical audio frequency, 1000 Hz for example, the acoustic wavelength in a water filled diffraction cell is 146 cm, which is too long to be included in any reasonable optical aperture. A spectrum analysis of an audio frequency can be performed in real time if the audio frequency is time compressed (or frequency multiplied) before being fed into the Debye–Sears analyzer. A spectrum analyzer is described in which the audio signal has been time compressed by a factor of 154 before analysis. The resultant bandwidth is limited by the length of the sample stored in the DELTIC frequency multipliers to a band of 88 Hz. Motion pictures will be shown of the analyzer output with various input signals, including speech.

LL5. Impedance at Mouth of Organ Pipe. S. A. ELDER AND W. E. FASNACHT (nonmember), *United States Naval Academy, Annapolis, Maryland 21402.*—Using velocity and pressure probes, the complex impedances at the mouths of closed and open organ pipes have been measured just inside the mouth and just outside, on either side of the jet. Outside the pressure leads the velocity by roughly 90° at each harmonic frequency and impedance magnitude increases approximately proportional to frequency as expected from linear end correction theory. Inside, the fundamental component of pressure is only a few degrees ahead of velocity in phase, while the higher harmonic components have a somewhat larger phase difference. This is in qualitative agreement with results of Ingard for nonlinear orifice impedance made for similar range of parameters. Fourier analysis of jet driving-pulse phase indicates that discontinuous overblow of closed pipe as contrasted with relatively smooth overblow of open pipe is due to sudden increase in "active" impedance of the jet associated with critical amplitude of third harmonic at which jet begins to enter pipe twice in a cycle. [Work supported by ONR.]

LL6. Search-Tone Measurements in Blown Wind Instruments. A. H. BENADE AND W. W. WORMAN (nonmember), *Case Western Reserve University, Cleveland, Ohio.*—A variable-frequency search tone is injected into an instrument's air column, and the response is detected by a probe microphone and narrow-band voltmeter tuned to the search-tone frequency. Accurate response curves are obtained at all frequencies except for narrow regions surrounding each of the strong (discrete) spectrum lines produced by the instrument as it plays. The search tone can be weak, or strong enough to produce nonlinear (heterodyne and synchronism) effects in the instrument's regeneration mechanism. The "blown" resonance peak for a bottle rises above the "unblown" natural frequency as the exciting air-jet velocity increases, but the *playing* frequency rises at a different rate. The width of the resonance is also somewhat sensitive to jet velocity. The first-mode resonance of a diapason organ pipe behaves similarly, but the higher "blown" modes show little change. There is considerable nonlinearity found in the regeneration mechanism of the organ pipe, but little in that of the bottle.

LL7. Development and Evaluation of Methods for Measuring the Characteristics of Small Acoustic Filters. WILLIAM S. GATLEY (nonmember), *The University of Missouri at Rolla, Rolla, Missouri* AND RAYMOND COHEN, *Purdue University, Lafayette, Indiana*.—Experimental methods are needed for determining the characteristics of small acoustic filters used in systems with pulsating gas flows, so that the performance of proposed filter designs in a particular system can be predicted according to plane-wave acoustic theory. Dependence on trial-and-error experimentation in solving noise control problems would thus be minimized. A literature survey revealed relatively little information useful for evaluating the acoustic performance of small filters and filter elements. Three methods for determining reflection and transmission factors are described, evaluated, and compared. A method employing a standing-wave tube of unique design is recommended for determination of reflection factors. Transmission factors are obtained from reflection factors and pressure measurements at the filter inlet and outlet. Use of an anechoic termination simplifies the calculations and increases accuracy; design and evaluation of such a termination is described.

LL8. A Schlieren System for Acoustic Measurements. J. V. SANDERS, *Naval Postgraduate School, Monterey, California 93940*.—Acoustic measurements with photographic-schlieren techniques are limited to regions of strong density gradients such as those found in shock waves and at the leading edge of intense sound waves. To study waves of less-than-finite amplitudes, a system employing photomultiplier-schlieren techniques was constructed. With this system, acoustic pressures as low as 20 dyn/cm^2 at 6 kHz were measured in air. A unique feature of the design is that the knife edge is vibrated at 300 Hz. This permits the absolute value of the pressure to be determined from known constants and a few length measurements. Microphone calibrations obtained using values of the pressure determined with this equipment agree with pistonphone calibrations to within ± 1 dB.

LL9. Acoustical Impedance of Thin Diaphragms. JOSEF MERHAUT, *Prague, Czechoslovakia*.—Differential equation describing the deflection η of a thin circular diaphragm, loaded with a complex specific acoustical impedance per unit area is solved. Then the total acoustical impedance Z_a of the diaphragm defined by $1/Z_a = j\omega \int\int \eta dS/p$ is calculated, where p is the acoustic pressure, and dS element of area. After expanding the Bessel functions in the result into series, zeroes and poles of Z_a are found. Further, a complete analog network for a thin diaphragm is given, including the formulas for the elements of this network.

LL10. On the Generation and Measurement of Acoustic Fields in Low Density Media. ASWINI KUMAR MOHANTY (nonmember), AND CALVIN C. OLIVER (nonmember), *School of Mechanical Engineering Purdue University*.—The results of experimental studies of generation and measurement of acoustic fields in argon, nitrogen and air at room temperature and at pressures as low as 30 mm Hg absolute will be presented. The motivation for the investigation was based on observations from experiments on microwave ionization of low pressure gases. The problems of transmission and detection of pressure perturbations in low-density media have implications, however, for other areas of investigation. A notable example occurs in the development of the necessary technology in space programs to probe physically low-density planetary atmospheres. Data are given in graphical form for SPL in a half inch tube resonated at 2000 Hz as a function of power, pressure, and gaseous medium. For example, in argon with a power input of 0.3 W to the acoustic driver, the SPL was reduced from 148 dB at atmospheric pressure to 90 dB at 30 mm Hg. The effects of the parameters of the study on electromechanical conversion efficiency will be indicated. Data will also be given on the manner in which the parameters affect attenuation in a damped probe. Finally, additional studies that are required to completely explain the results of the present investigation will be outlined.

LL11. General Evaluation of Sonar Transducers by Average Time Holographic Interferometry. RALPH M. GRANT (nonmember) AND CHARLES F. JACOBSON (nonmember), *G-C Optronics, Inc., Ann Arbor, Michigan*.—A brief explanation of the average time holographic interferometry technique applied to the evaluation of sonar transducers [R. M. Grant and W. Von Winkle, "Interferometric Holography as a Tool for the Study of Flexures in Active Sonar Transducers," paper invited and presented at the First Intern. Conf. Laser App. Paris, France (July 1967)] will be presented. In addition to a brief theoretical discussion of the average time technique, examples will be given of the application of the above technique to both design and nondestructive testing of sonar transducers. Results will be given for both single elements and arrays.

LL12. Experimental Results of Optical Holographic Testing of a Sonar Transducer. CHARLES F. JACOBSON (nonmember), AND RALPH M. GRANT (nonmember), *G-C Optronics, Inc., Ann Arbor, Michigan*.—A brief review of Powell and Stetson's original work on vibration holography [R. Powell and K. Stetson, J. Opt. Soc. Am. 55, 12 (1965)] will be given and the mathematics discussed for determining the relative displacement of the transducer surface from the vibration hologram. Experimental results of tests of a prototype sonar transducer (several models) using holographic interferometry are given. Hologram vibration patterns at various drive frequencies will be shown and corresponding displacement or flexure diagrams shown. Experimental comparison of unloaded (in air) and loaded (in water) operation [R. M. Grant *et al.*, "Underwater Holography," J. Opt. Soc. Am. 56, 8, 1142 (1966)] will also be given. The feasibility of predicting performance under loaded operation from unloaded holographic measurements will be discussed.

LL13. An Underwater-Sound Level Meter. J. J. TRUCHARD, *Defense Research Laboratory, The University of Texas at Austin, Austin, Texas 78712*.—The design of an audio-frequency underwater-sound level meter is presented. It consists of a hydrophone, low-noise preamplifiers, and a true rms voltmeter. The hydrophone consists of two ceramic hemispherical shells cemented together with an especially designed 40-dB gain preamplifier inside the hemispheres. The use of a preamplifier inside the ceramic element permits the use of short leads between the ceramic and the preamplifier without a separate preamplifier housing. The self-noise level of the preamplifier is more than 20 dB below the noise level equivalent to sea state zero at frequencies up to 1000 Hz. The thermal noise of the hydrophone is compared to the theoretical thermal-noise limit of a unit-efficiency, omnidirectional hydrophone. The square-law detector in the true rms meter circuit uses a field-effect transistor for squaring. The frequency range of the underwater-sound level meter is 20–20 000 Hz. All components except the hydrophone are useful to 100 kHz. The underwater-sound level meter was used to measure the noise level under extremely quiet conditions in an anechoic Mason tank. The noise levels measured were as low as 17 dB below equivalent sea-state-zero noise at 1000 Hz. [This work was sponsored by the U. S. Naval Ship Systems Command.]