PROGRAM OF

The 113th Meeting of the Acoustical Society of America

Hyatt Regency Hotel • Indianapolis, Indiana • 11–15 May 1987

MONDAY EVENING, 11 MAY 1987 — CELEBRATION HALL, 7:00 TO 9:00 P.M.

Tutorial Lecture

Digital signal processing. Lawrence R. Rabiner (AT&T Bell Laboratories, Speech Research Department, Room 2D-533, Murray Hill, NJ 07974)

In all fields of acoustics, the response of systems of interest involves the analysis and the processing of a signal. This signal can be the radiated acoustic pressure from a hydrophone, the acceleration of a mass due to local acoustic excitation, the speech of a spoken word, or the melodious pattern of a music instrument. In each of these cases the signal has to be processed and analyzed. The field of digital signal processing is one area that has been attracting more interest over the last few years. In this tutorial the field of digital signal processing (DSP) will be explained together with the theory of linear systems, sampling, and spectrum analysis. Comparisons between digital and analog processing will be discussed. Other topics that will be presented are discrete Fourier series, sampling, aliasing, FIR and IIR filters, spectrum analysis, and fast Fourier transform. The session will conclude with detailed examples of applications of digital systems in different areas of interest such as communication in general and speech processing.

TUESDAY MORNING, 12 MAY 1987 — REGENCY BALLROOM A & B, 8:15 TO 11:54 A.M.

Session A. Speech Communication I: Speech Perception

Robert A. Fox, Chairman

Department of Speech and Hearing Science, Ohio State University, Columbus, Ohio 43210

Chairman's Introduction—8:15

Contributed Papers

8:20

A1. The problem of serial order in auditory word recognition. Howard C. Nusbaum (Department of Behavioral Sciences, The University of Chicago, 5848 S. University Avenue, Chicago, IL 60637), Steven L. Greenspan (AT&T Bell Laboratories, Naperville, IL 60566), and Mathew Jensen (Department of Behavioral Sciences, The University of Chicago, Chicago, IL 60637)

Recently, there has been a resurgence of interest in parallel distributed processing models of human perception. When speech perception is modeled in this type of spatially distributed network, a problem arises in coding the temporal order of perceptual units such as phonemes or words. In general, three solutions to this problem have been proposed: First, perceptual units may be context coded such as in context-sensitive allophones. The order of units presented at different points in time can be determined by matching the context "edges" of each activated unit. Second, different portions of the network may represent different time frames. By this approach, the recognition of each successive perceptual unit activates representations in successive segments of the network. Finally, temporal order may be represented in the computational dynamics of the network. In this case, expectations about serial order are used to shift the focus of processing attention within the network. Thus, while the first two approaches recode temporal order into a spatial or spatial-like representation, the third uses a temporal representation. Each of these approaches has positive and negative attributes, the implications of which will be discussed for a neuromorphic theory of speech perception. [Work supported, in part, by NIH.]

8:32

A2. The neighborhood activation model of auditory word recognition. Paul A. Luce (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

The neighborhood activation model (NAM) of auditory word recognition describes the processes by which a stimulus word is identified in the context of phonetically similar words activated in memory. Stimulus input activates a set of acoustic–phonetic patterns in memory that must be discriminated and chosen among. These acoustic–phonetic patterns receive activation levels proportional to their similarities to the stimulus input. The activation levels may then be adjusted by biases arising from higher-level information, such as word frequency. The interaction of the bottom-up sensory input and top-down biasing information is assumed to take place within individual processing units called word decision units. These units monitor the activation levels of their acoustic–phonetic patterns, any higher-level information that may optimize decisions among the competing patterns, and the activity of all other word decision units.

The NAM states that increasing the number of acoustic–phonetic patterns activated in memory by the stimulus input will slow processing and reduce identification accuracy. It also states that effects of word frequency are directly tied to the number and nature of similar words activated in memory and that word frequency is not intrinsic to the activation levels of the acoustic–phonetic patterns. [Work supported by NIH Grant NS-12179.]

8:44

A3. Auditory word recognition is not more sensitive to word-initial than to word-final stimulus information. M. J. van der Vlugt and S. G. Nooteboom (Institute for Perception Research, Eindhoven, The Netherlands)

Several accounts of the human recognition of spoken words assign special importance to stimulus-word onsets. The experiment described here was designed to find out whether such a word-beginning superiority effect is due to a special sensitivity of the word recognition process to word-initial stimulus information, or rather to the special importance of word onsets for the proper alignment of stimuli with word candidates. Twenty-eight polysyllabic monomorphematic test words were selected, all having the special characteristic that their left-to-right and right-to-left recognition points coincided in the same phoneme. These words were synthesized from diphones, and, of each word, four versions were prepared: (a) without noise; (b) with noise masking segment perception in the word-initial fragment; (c) with noise masking segment perception in the word-final fragment; and (d) with noise masking the whole word. These stimuli were, appropriately blocked, distributed over four stimulus tapes, and presented to four groups of 12 subjects for word recognition. Percentages correctly recognized real words were, for the four conditions: (a) 89%, (b) 54%, (c) 59%, and (d) 13%. The difference between (b) and (c) was not significant. This is interpreted as evidence that (1) auditory word recognition is not more sensitive to word-initial than to word-final stimulus information, and (2) stimulus information from all parts of a spoken word can be used efficiently, as long as its proper time alignment with word candidates in the mind of the listener is ensured.

8:56

A4. Effects of talker uncertainty I: Auditory word recognition. John W. Mullennix, David B. Pisoni, and Christopher S. Martin (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

The production and resulting acoustic composition of spoken words vary as functions of individual talker characteristics. However, the effects of talker differences on auditory word recognition processes have been largely ignored by researchers working in speech perception. In the present study, the effects of talker differences on perception of spoken words were examined by manipulating two factors: talker uncertainty (i.e., words produced from a single talker or from 15 talkers) and lexical structure (i.e., high- or low-density lexical items). In the first experiment, identification performance for words presented in noise was worse when the items were produced by different talkers, than by a single talker. In the second experiment, latencies and accuracy in a naming task were also worse under multiple-talker condition. In both experiments, lexical density alone did not have a significant effect on performance. The relationship of these results to the underlying processing operations involved in auditory word recognition will be discussed. [Work supported by NIH.]

9:08

A5. Effects of talker uncertainty II: Encoding speech into memory. Christopher S. Martin and John W. Mullennix (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Individual talkers vary in a large number of articulatory and acoustic characteristics. Little research has been conducted concerning the effects of talker uncertainty on encoding speech into memory. In order to examine this problem, a serial-ordered recall experiment was conducted. The stimuli were CVC monosyllabic English words produced by different male and female talkers. Listeners were presented with lists of ten items produced by a singler talker or lists of ten items produced by multiple talkers, and were required to recall the items in the order presented. Percent correct recall in the primary region of the serial position curve was lower as the amount of talker variability within the lists increased. The results suggest that the encoding of speech into memory is affected by uncertainty from trial to trial due to talker variability. The hypothesis that perceptual processes involved in talker normalization are located at an auditory-to-phonetic level will be discussed. [Work supported by NIH.]

9:20

A6. The lexical status effect on place of articulation and voice onset time contrasts. Marjorie A. Reed (Department of Psychology, Cleveland State University, Cleveland, OH 44115)

Ganong [J. Exp. Psychol.: Hum. Percept. Perform. **6**, 110–125 (1980)] first reported that the presence of a word at one end of a voice onset time series influenced the classification of stimuli near the boundary. Fox [Percept. Psychophys. **34**, 526–540 (1983)] reported a similar effect using place of articulation stimuli. In the present experiment, the lexical status effect on two voice onset time continuua (/bot/ to /pot/ and /bok/ to /pok/) was compared to that on two place of articulation continuua (/bon/ to /don/ and /bop/ to /dop/). Results showed effects of lexical status for both types of stimuli. Ambiguous tokens near the category boundary were perceived in favor of the word, while the classification of endpoint stimuli was not influenced. The lexical status effect was significantly larger for voice onset time stimuli than for place of articulation stimuli. This difference may result from the time required for the lexical status effect to develop, suggesting alternate interpretation of Fox's data on the time course of the effect. Implications for interactive and separate stage models are discussed.

9:32

A7. Context effects in spoken language. Robert Pedlow (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

The effects of different types of linguistic context on the intelligibility of spoken language were examined. A set of words was produced by the same speakers in three different types of linguistic context: ordinary sentences, anomalous sentences, and a word list context. The anomalous sentences were constructed by changing one nonadjacent-to-target word to an anomalous alternative. The word list context was set up as short, sentencelike strings, seven words in length, including the target word. In the production phase of the experiment, speakers were instructed to speak with the same intonation over all the materials. To encourage this, the different context types were randomly interleaved in the materials which speakers read. No significant differences were found in the intelligibility of the words produced in the different context conditions. It is argued that these results demonstrate that, contrary to the findings of (Lieberman, 1963), it is the overall speech "mode," rather than the immediate linguistic context, which acts to determine the intelligibility with which individual words are produced.

9:44

A8. Perceptual differentiation of spontaneous and read utterances after resynthesis with monotone fundamental frequency. R. E. Remez, R. S. Bressel (Department of Psychology, Barnard College, 3009 Broadway, New York, NY 10027), P. E. Rubin, and N. Ren (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Our prior research revealed that naive listeners can differentiate spontaneous speech and read speech even when lexical, syntactic, and semantic components are equated in the two modes of production. On a typical trial in our tests, a listener hears a pair of excerpts identical in these three levels of description, taken from a spontaneous and a read monologue produced by the same talker. The perceptual task is to identify the spontaneously produced member of the pair. In the present case, LPC resynthesis was employed to create constant F_0 versions of highly differentiable pairs of sentences, to test our working hypothesis that the variation of F_0 serves as the most reliable index of perceptual performance. Three conditions were tested: whole sentence comparisons, initial-half-sentence comparisons, and final-half-sentence comparisons. Subjects performed above chance when listening to whole sentences only, indicating the existence of residual information for production mode in the absence of F_0 variation. Implications and extensions of this finding will be discussed. [Work supported by NINCDS and NICHHD.]

9:56–10:06
Break

10:06

A9. Forward recognition masking: Another look at the "Echo." Gail R. Tomiak (Department of Psychology, Park Hall, State University of New York at Buffalo, Buffalo, NY 14260)

Auditory backward recognition masking effects have been interpreted as reflecting the existence of an echoic memory: an unencoded sensory store whose contents decay rapidly and are easily overwritten by succeeding auditory stimulation. However, it is equally likely that masking effects reflect the time course of an early stage of auditory processing. To distinguish these alternative conceptualizations, vowel stimuli were used in a forward recognition masking procedure. The interstimulus interval (ISI) remained constant, while the mask duration (and hence the stimulus-onset asynchrony, SOA) was varied. The simple storage model predicts that the efficacy of a mask is a function of the time between mask offset and target onset (ISI). Since this interval was constant, equal masking effects would be expected at all SOA intervals. The perceptual processing view predicts that target recognition performance would improve with increasing mask duration (increasing SOA). As mask duration/SOA increases, it is likely that processing of the mask would be completed before target onset. The results, and the implications of these findings, will be discussed. [Work supported by NINCDS.]

10:18

A10. Release burst masking effects for temporal order identification. Richard E. Pastore, Jody K. Layer, Robert J. Logan, and Crystle B. Morris (Department of Psychology, SUNY University Center, Binghamton, NY 13901)

Nearly three decades ago, Hirsh [J. Acoust. Soc. Am. **31**, 759–761 (1959)] found that the threshold for identification of temporal order of onset is significantly longer than the threshold for detection of onset asynchrony, and proposed that the order identification threshold might serve as a basis for the contrast between voiced and unvoiced aspirated stop consonants in initial position. In the last decade, a number of researchers have argued that temporal order identification threshold does not play a significant role in voicing contrast. One basis for this argument against a phonetic perception role is that the 15- to 20-ms range of typical order identification thresholds for simple stimuli are significantly shorter than VOT boundaries typical for synthetic stop consonants which include an initial release burst. The current research investigated the possible contribution of an initial noise burst analog as a masking stimulus. The initial noise burst shifts temporal order identification thresholds to significantly longer onset asynchronies which are equivalent in magnitude to VOT voicing contrast boundaries.

10:30

A11. Perception of onset bursts in /ba da ga/ syllables by good and poor readers. Judith A. Parker (Department of Cognitive and Linguistic Sciences, Brown University, Providence, RI 02912)

In identification and discrimination tasks, good and poor readers listened to tapes of synthesized stimuli consisting of 350-ms full (burst + formant transition) and partial (formant transition only) acoustic signals for voiced stop CV syllables. A free identification procedure was used to elicit percepts and an oral response method was used to avoid possible reading/writing confounds. The ANOVA, probit analysis, and signal detection analysis were used to evaluate findings. As a group, poor readers demonstrated significantly weaker categorical perception for the /da ga/ boundary in stimuli lacking bursts. Burstless signals produced severe problems for some poor readers; others performed similarly to the higher averaged performances of the good readers. Individual differences were noted: impaired perception regarding place, manner, or boundaries occurred in the full-stimulus condition, and more often for the partial condition. The data are relevant to theories that propose a hierarchy of acoustic cues for the perception of linguistic contrasts and also suggest a possible perceptual deficit that may be involved in some forms of dyslexia.

10:42

A12. Effects of preceding context on perception of voice onset time. Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

When discriminating pairs of stimuli from an acoustic voice-onset-time (VOT) continuum, English-speaking listeners typically show a performance peak in the region of the phonetic category boundary. An ongoing series of experiments demonstrates that the location and height of this peak are affected by preceding phonetic context. While any preceding context appears to lower discrimination performance, perhaps by interfering with auditory memory, a preceding [s] also shifts the peak towards the short-VOT end of the continuum and, correspondingly, increases "voiceless" responses in a labeling task. This latter effect (trivial within a syllable because of English spelling conventions) occurs even when a word boundary intervenes and does not seem to be mediated by the duration of the silent closure interval following the [s]. A hypothesis currently being pursued is that phonological voicing decisions for word-initial stop consonants are sensitive to the voicing status of preceding phonetic segments. [Work supported by NICHD.]

10:54

A13. Evaluation of the constant ratio rule for consonant confusions. Theodore S. Bell, Donald D. Dirks, and Gail E. Kincaid (UCLA School of Medicine, CHS 62-142, Los Angeles, CA 90024)

The constant-ratio rule for consonant confusions asserts that the ratio of errors of any consonant to any others remains constant regardless of the size of the matrix from which the cells were chosen. In this study, log-linear models were applied to test the constant-ratio rule. Normal-hearing young adults were presented various closed-set arrangements of syllable tokens, spoken by a male and female speaker, and selected from a set of 14 VCs each paired with the vowel /ɑ/. Stop consonants and fricatives, both voiced and voiceless, were presented at three presentation levels as isolated sets (by feature) and in combination. The relationships among voiceless stop consonants remained constant irrespective of the size and nature of the response foil. Similarly, voiceless fricatives were unaffected by the nature of the closed set. In contrast, patterns of errors among voiced stops were dependent on the set of alternatives, as were voiced fricatives. Speaker differences, individual differences among listeners, and implications relating to the generalizability of confusion data collected in small closed-set arrangements are also discussed. [Work supported by a grant from NINCDS.]

11:06

A14. Effects of message redundancy on the calculations of speech intelligibility. Chaslav V. Pavlovic (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

The articulation index methods used today assume that the distribution over frequency of the usable information content of speech does not depend on various sequential or contextual constraints existing in the message. This assumption has repeatedly been shown to be inaccurate, but the suggested alternative approaches reduce the generality of the theory. They require a separate frequency importance function for each degree of redundancy. In this study, the frequency importance function for the material of average everyday redundancy is derived. The function is flat on the Bark scale. This suggests that the speech code is optimally matched to the receiver (auditory system). The optimization refers to maximizing the reception rate of information.

11:18

A15. Response time to a sentence verification task as a function of LPC narrow-band processing and bit error rate. A. Schmidt-Nielsen and Howard J. Kallman (Naval Research Laboratory, Code 7520, Washington, DC 20375)

The comprehension of narrow-band digital speech with bit errors was tested using a sentence verification task. The difficulty of the verification task was varied by using predicates that were either strongly or weakly related to the subjects. (A toad has warts./A toad has eyes.) In addition, half of the sentences were about category relationships, and half were about property relationships. (A salmon is a bird./A camel has horns.) The test conditions included unprocessed speech and speech processed using a 2400 bit/s linear predictive coding (LPC) voice processing algorithm with random bit error rates of 0%, 2%, and 5%. In general, response accuracy decreased and reaction time increased with LPC processing and with increasing bit error rates. False sentences took longer to verify than did true sentences, but there were fewer errors for the false sentences. Weakly related true sentences and strongly related false sentences were more difficult than their counterparts. Interactions between sentence type and processing conditions will be discussed.

11:30

A16. Perception of natural and vocoded sentences among English monolinguals and German-English bilinguals. Molly Mack (Division of English as an International Language and Department of Linguistics, University of Illinois, Urbana, IL 61801)

In the present study, the perceptual performance of 24 English monolinguals and 24 German-dominant German-English bilinguals was subjected to detailed analysis. Stimuli consisted of 57 semantically anomalous natural and vocoded English sentences. Results revealed that the monolinguals made an average of 5.92 errors and the bilinguals 66.58 errors in the natural condition, while the monolinguals made an average of 30.58 errors and the bilinguals 83.17 errors in the vocoded condition. For both groups, phonemic errors predominated, although approximately 30%-40% of the errors were morphosyntactic and/or lexicosemantic. {These latter results were essentially in agreement with previous findings [M. Mack and B. Gold, MIT Lincoln Laboratory Tech. Rep. 703 (1985) and J. Acoust. Soc. Am. Suppl. 1 77, S10-S11 (1985)].} Further, the bilinguals' responses to the vocoded stimuli suggested that they were employing a theory-driven rather than a data-driven response strategy, presumably due to the difficulty of the task. [Work sponsored by the Department of the Air Force.]

11:42

A17. Listener-talker interaction: Is there an "autophonetic" effect? John W. Hawks and James D. Miller (Central Institute for the Deaf, 818 South Euclid Avenue, St. Louis, MO 63110)

Based on recent physiological evidence for neural selectivity to self-produced song in the auditory system of the White-Crowned Sparrow, a hypothesis stating that the human perceptual mechanism for speech may be most acute for self-produced speech sounds was tested. A test of the ability to identify monosyllabic words was presented in a noise background. The same subjects served as both talkers and listeners. A small, but statistically nonsignificant, advantage is found when listener and talker are the same person. Additionally, while variation between talkers was nearly twice the variation between listeners, the performance of a particular talker-listener pair could be predicted by a simple equation. This equation gives equal weight to the talker's and listener's average performance, and thus the talkers, by virtue of their greater variance, tend to have a greater influence. [Work supported by NIH.]

TUESDAY MORNING, 12 MAY 1987 — MT. RAINIER, 8:30 TO 11:35 A.M.

Session B. Noise I and Psychological and Physiological Acoustics I: History and Development of Hearing Protection Devices

Elliott H. Berger, Chairman
E-A-R Division, 7911 Zionsville Road, P. O. Box 68898, Indianapolis, Indiana 46268-0898

Chairman's Introduction—8:30

Invited Papers

8:35

B1. History and development of hearing protection devices. W. Ian Acton (ISVR, University, Southampton, S09 5NH, United Kingdom)

Military need has spurred the invention of hearing protectors since the times of Greek mythology. Earplugs were patented in 1864 and canal caps attached to an adjustable headband in 1884 as protection for soldiers and sailors. Attempts to limit gunfire noise by mechanical devices commenced in 1905, leading to Mallock-Armstrong plugs for use in the First World War. Disposable earplugs were patented in 1914 also. Systematic development of earplugs for forces during the Second World War culminated in the V-51R. Leather flaps over the ears were supposed to protect the crews of military aircraft. The increased noise of jet engines led to the production of recognizable earmuffs, although the stiffness of the cushions required a strong headband to ensure a seal against the head. This caused a vicelike grip. The problem was overcome with the development of fluid-filled cushions by Shaw in 1954. Progress has continued towards comfort and acceptability, with glass-down appearing in the late 1950's, conformable foam plugs in the 1970's, and nonlinear electronic systems incorporated into muffs in the 1980's.

9:05

B2. Analytical concepts and models for calculating hearing protector attenuation. Edgar A. G. Shaw (Division of Physics, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

In principle, the ideal hearing protector is a rigid airtight cover or plug that encloses a volume of air in contact with the eardrum and is sealed to an immovable object (the head or the external ear) by a resilient airtight cushion or interface. According to this simple view, the sound attenuation curve is solely dependent on a few lumped elements: the mass and area of the cover or plug, the acoustic impedance presented by the enclosed space, and the stiffness and mechanical resistance of the cushion or interface. In practice, the attenuation is also dependent on air leakage, body-conducted sound, the lack of rigid support, wave effects, and mechanical imperfections. These factors and others have been measured, estimated, and modeled with considerable success though perfect agreement between theory and experiment remains elusive. So, perhaps there is still room for further work of a fundamental nature. In the meantime, despite their limitations, the classical analytical models continue to provide an invaluable foundation for hearing protector design.

9:35

B3. Development of a unique passive level-dependent hearing protector. C. H. Allen (The Clayton H. Allen Corporation, 80 South Road, Chebeague Island, ME 04017-9710) and E. H. Berger (E-A-R Division, Cabot Corporation, P. O. Box 68898, Indianapolis, IN 46268-0898)

The need for protection of hearing against intermittent high-level sounds and the opposite need for retaining hearing sensitivity during low-level intervals, have led to the development of level-dependent, commonly called "nonlinear," hearing protectors. This paper deals with the design and refinement of passive means for automatically and instantaneously altering the amount of attenuation, provided at the ear, in response to the incident sound level. The early successful development of a nonlinear earplug for protection against gunfire is followed through laboratory and field tests, and the advantages and limitations of such an earplug are discussed. The more recent application of a nonlinear orifice to earmuffs, for various applications, introduces several new difficulties. Their identification and the means that have been found suitable for use in practical level-dependent earmuffs are discussed. Test data are presented showing acoustic characteristics over a range of incident sound levels from 100 dB, or lower, to 160 dB, or higher. Results obtained with steady state, broadband, and narrow-band noise are compared and are shown to be similar to results obtained with impulse noise, such as experienced, for example, from gunshots or forging hammers.

10:05

B4. History and development of active noise reduction hearing protection. Richard L. McKinley (Harry G. Armstrong Aerospace Medical Research Laboratory, Wright–Patterson Air Force Base, OH 45433)

Ever since Lueg's 1936 patent, the application of active noise reduction technology to hearing protection has been a goal of numerous research groups. This paper will describe the various efforts to apply active noise reduction technology to earcups over the time span from 1936 through the present. Each of the various efforts will be described in terms of basic cancellation technique, hardware realization, and both laboratory and field performance when available. In addition, a working model of the current active noise reduction headset developed jointly by the Air Force and Bose Corporation, similar to the unit used on the around the world flight of the Voyager, will be presented.

10:35

B5. Noise attenuating earphones for audiometric testing. Mead C. Killion (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007) and Elliott H. Berger (E-A-R Division, 7911 Zionsville Road, Indianapolis, IN 46268)

Tests of the noise excluding properties of the TDH-50P/MX-41AR, the Audiocup, and the ER-3A insert earphone were performed in a diffuse-field facility complying with ANSI S12.6-1984. Data on attenuation were obtained monaurally with the nontest ear plugged and muffed. Results generally agreed well with previously reported measurements. A broadband noise, shaped by a multifilter, allowed a direct test of the ANSI S3.1-1977 permissible background noise levels for testing to audiometric zero under TDH-39/MX-41AR headphones. This "ANSI noise" raised the average thresholds of 15 normal-hearing test subjects by 3 to 5 dB at the 500- to 4000-Hz octave frequencies. With a noise shaped to the less stringent OSHA-1983 regulation, average thresholds were raised 9 to 17 dB. Introduction of an "ENT office noise" with 55 dB(A) overall level raised average thresholds at those frequencies by 11, 5, 1, and 0 dB with the Audiocups and less than 2 dB with the recommended fully ("deeply") inserted ER-3A eartips. Measured threshold elevations agreed closely with predictions based on a critical ratio calculation utilizing actual sound field noise levels and measured attenuations. A conservative rule of thumb for testing to audiometric zero with the ER-3A earphone appears to be an office noise level of 45 dB(A) or less.

Contributed Papers

11:05

B6. Hearing protector attenuation for impulse noise. S. E. Forshaw (Defence and Civil Institute of Environmental Medicine, P. O. Box 2000, Downsview, Ontario M3M 3B9, Canada)

The study of the effectiveness of hearing protection for impulse noise poses several problems. Due to the transient nature of the noise, real-ear-at-threshold (REAT) data are not readily applicable since current impulse-noise criteria are based on either (1) peak pressure, some measure of impulse duration, and the number of impulses, or (2) total A-weighted energy. It is possible to measure the effect of an earmuff on these parameters by positioning a miniature transducer under the muff at the wearer's ear canal. An alternative arrangement which permits the study of both ear muffs and plugs involves the use of an acoustic test fixture. The test fixture must simulate those characteristics of the head and outer and middle ears that affect the attenuation of hearing protectors on real ears. These include the viscoelastic characteristics of the circumaural and intra-aural skin, the inertial properties of the head, the frequency response of the auditory canal, and the mechanisms of bone conduction and physiological masking. DCIEM has sponsored by contract the development of such a test fixture [C. Giguere *et al.*, J. Acoust. Soc. Am. Suppl. 1 78, S5 (1985)]. Attenuation results with the test fixture compare reasonably well with REAT-derived data. Agreement between the two methods could be improved for earplugs if the thickness of the artificial intra-aural skin were reduced. On-going research is directed towards studying human intra-aural skin, and measuring the peak attenuation, waveform-duration increase, and linear and A-weighted energy reduction of protectors for short- and long-duration impulses.

11:20

B7. The development of hearing protective systems for use in the high-noise environments of army combat vehicles. Mitchell S. Mayer and Tat Y. Fung (U.S. Army Communications Electronics Command, AMSEL-RD-COM-TR-3, Fort Monmouth, NJ 07703-5202)

Military vehicles are designed for maximum capabilities in areas of performance, armament, and endurance. Acoustic noise suppression at the source is not considered in the system design because of its degradation on performance, as well as its cost impact. Therefore, it is left to the crew helmet to provide for the hearing protection of the crew member. In order to protect the hearing and improve the job performance of the crew, there are two major noise paths to be considered: first is the noise pickup by the communication microphone, along with the speech communication; second is the environmental noise that penetrates the earcup. It is believed that additional improvements to such devices as noise canceling microphones and earcup/transducer assemblies have reached their practical limits. Therefore, research efforts are presently being conducted to incorporate active speech processing for rejecting noise picked up by the communication microphone, and active noise reduction to reduce the environmental noise penetrating the earcup. These efforts will be discussed, along with new types of test equipment and evaluation procedures that have been developed to more accurately evaluate existing communication components and future active speech processing/noise reduction systems.

TUESDAY MORNING, 12 MAY 1987 REGENCY BALLROOM C, 8:30 A.M. TO 12:00 NOON

Session C. Psychological and Physiological Acoustics II: Cochlea, VIII Nerve, and Evoked Responses

Donald Wong, Chairman
Indiana University School of Medicine, Medical Science 258, 635 Brownhill Drive, Indianapolis, Indiana 46223

Contributed Papers

8:30

C1. Tectorial membrane: Static mechanical properties *in vivo*. Jozef J. Zwislocki, Steven C. Chamberlain, and Norma B. Slepecky (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-5290)

In a mammalian cochlea, stimulation of the hair cells must be associated with shear motion between the organ of Corti and the tectorial membrane. Knowledge of the mechanical properties of the tectorial membrane in its natural environment is essential for our understanding of this process. Recently, the present authors have been able to visualize the tectorial membrane in live Mongolian gerbils by fenestrating the scala media of the second cochlear turn without disturbing Reissner's membrane. The anatomical configuration of the cochlea in these animals allowed us to observe the tectorial membrane over its entire width. A micropipette was inserted into the membrane and moved radially and longitudinally approximately parallel to the membrane surface. When vitally stained with Janus green, the membrane showed astonishing toughness and resilience in both directions. Its deformation patterns resembled those of a rubber band. When vitally stained with Alcian blue, its tensile strength was decreased, and it tore rather easily. Subsequent microscopic investigation of the membrane *in vitro* showed that the two stains bind to different structures.

8:45

C2. Perilymph volume flow measured under physiological conditions. Kenji Ohyama, Alex N. Salt, and Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

This study made use of an ionic tracer (trimethylphenylammonium, TMPA) to mark perilymph in one region of the cochlea. Perilymph flow was assessed by monitoring tracer movement to other regions with ion selective electrodes. Small, nontoxic, quantities of tracer (typically a 50-nl bolus of 150 nM TMPACl) were required, since concentrations as low as 1 μM were readily detected. Stringent precautions were taken to seal the injection and recording electrodes into the perilymphatic scalae so that no artifactual flow was induced by perilymph leakage. Results were compared with a mathematical model of tracer dispersion by volume flow and passive diffusion. For scala tympani, TMPA dispersion corresponded to a flow rate of 1.6 nl/m in the apical direction. This rate was significant ($p < 0.05$), although the physiological effects of such a low flow rate are presumed to be negligible. Perforation of the cochlear apex resulted in flow rates of over 1 μl/m, almost 1000 times the physiological rate. In scala vestibuli, no significant flow could be detected under normal conditions. These results demonstrate that in the sealed state, longitudinal volume flow of perilymph is extremely low. Artifactual volume flow is readily induced by procedures which involve perforation of the otic capsule. [This work supported by NIH.]

9:00

C3. Mechanism of endocochlear potential generation by stria vascularis. Alec N. Salt, Ivo Melichar,[a] and Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

It is widely assumed that the endocochlear potential (EP) is generated by electrogenic ion transport by the marginal cells of stria vascularis (SV). It has recently been reported [I. Melichar and J. F. Syka, Hear. Res. **25**, 35–43 (1986)] that a potential up to 10 mV higher than EP was recorded as microelectrodes were advanced through SV. Dye marking studies demonstrated that this potential is associated with the marginal cell region. The present authors have studied this potential further using small-tipped (0.5–2.0 μ o.d.), double-barreled potassium selective electrodes. As the electrodes were advanced through SV, a region of high K content with near zero potential was observed, corresponding to the basal cells. Immediately following, a region of positive potential (higher than the EP) and *low* K content was observed. In most cases endolymph was then reached, but sometimes two distinguishable regions of positive potential and high K were observed, corresponding to marginal cells and endolymph, respectively. The observation of a positively polarized low K region is not consistent with the rôle of the marginal cells as the generator of EP. On the basis of the observation that K is close to equilibrium between the basal cells and the extracellular space of stria, it is proposed that EP is generated by *passive* diffusion of K across this boundary. [This work supported by NIH.] [a] Also at Czechoslovak Academy of Sciences, Prague, Czechoslovakia.

9:15

C4. Cytochalasin D suppresses the compound action potential and summating potential of the guinea pig cochlea. S. E. Barron, R. P. Bobbin, P. S. Guth, and C. H. Norris (Department of Pharmacology, Tulane Medical School and Department of Otorhinolaryngology, LSU Medical School, New Orleans, LA 70112)

Although actin has been localized in hair cells, evidence for its role in intact cochlear preparations is lacking. The present study was undertaken to test the hypothesis that actin is involved in cochlear function (e.g., the active process) by observing the effects of a potent filamentous muscle actin inhibitor, cytochalasin D, on evoked cochlear potentials. Perilymphatic spaces of guinea pig cochlea were perfused with Ringer solutions containing either cytochalasin D (10^{-8} to 10^{-5} M) or DMSO (0.00005% to 0.05%) at a rate of 2.5 μl/min for 10 min [Bledsoe *et al.*, Hear. Res. **4**, 109 (1981)]. Immediately after each period of perfusion, the compound action potential (CAP), cochlear microphonics (CM), and summating potential (SP) evoked by 10-kHz tone bursts of varying intensity were recorded from an electrode in the basal turn. Cytochalasin D suppressed CAP and SP in a dose-dependent fashion but did not affect CM. DMSO had no effect. These results provide pharmacological evidence that actin is involved in cochlear function—possibly the active mechanical process of the organ of Corti. [Supported by NSF, NINCDS, the Southern Hearing and Speech Foundation, Kresge Foundation, and the LA Lions Eye Foundation.]

9:30

C5. Anatomical effects of intense tone stimulation in the goldfish ear: Dependence on sound-pressure level and frequency. Mardi Cox, Peter H. Rogers (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332), Arthur N. Popper, and William M. Saidel (Department of Anatomy, Schools of Medicine-Dentistry, Georgetown University, Washington, DC 20007)

The final results of an experimental investigation to study the extent of frequency regionalization in the ear of goldfish are presented. Goldfish about 6 in. in body length were subjected to intense tones at 250 and 500 Hz, and four different sound-pressure levels. They were placed in a waveguide and constrained as close as possible to a pressure antinode so that the primary response of the inner ear was due to the induced motion of the swimbladder and Weberian ossicles. Both saccular and lagenar maculae were examined under a scanning electron microscope to determine the location and extent of hair cell damage as a function of frequency and sound-pressure level. The results are not inconsistent with the gross frequency regionalization in the saccular macula of codfish determined by P. S. Enger [*Hearing and Sound Communication in Fishes* (Springer, New York, 1981), pp. 243–255]. In addition, the results indicate a possible breakdown of the Weberian apparatus at extremely high sound-pressure levels where the primary site of damage switches from the saccule to lagena. This is consistent with the behavior of the system based on its viscoelastic properties as postulated by R. McN. Alexander [J. Exp. Biol. **38**, 747–757 (1961)]. Work supported in part by ONR and NIH.]

9:45

C6. The effects of hypoxia and temperature change on tuning and rate suppression in goldfish auditory nerve fibers. R. Fay and T. Ream (Parmly Hearing Institute, Loyola University of Chicago, Chicago, IL 60626)

Goldfish auditory (saccular) nerve fibers are modestly tuned with best frequencies (BF) between 200 and 1500 Hz. Many fibers tuned below 400 Hz also exhibit rate suppression for signals well above BF. To investigate the origins of this frequency selectivity and suppression, the responsiveness of saccular fibers was measured during and after transient hypoxia, and during changes in body temperature of about ± 5 °C. Respirator water temperature controlled body temperature (monitored in the brain). In hypoxia studies, the respirator water was turned off for about 20 min. Low temperature and hypoxia had similar effects: a reduction in spontaneous activity, sensitivity, and responsiveness, and greater adaptation and suppression. Although temperature reduction and hypoxia caused a reduction in the most excitatory frequency at any given intensity (suggesting a lowering of the fiber's BF), the present authors could not clearly distinguish between these effects and those caused simply by reducing sound intensity. Thus, tuning is not strongly dependent on temperature or oxygen supply, and the mechanisms of suppression in the saccular nerve are likely different from those operating in the cochlear nerve of most other vertebrates. [Work supported by a Program Project Grant from NIH and NINCDS.]

10:00

C7. Otoacoustic emissions evoked by bone-conduction stimuli. Lu Yuanyuan and Zhang Qin (Deafness Research, Shanghai College of Chinese Traditional Medicine, 530 Lingling Road, Shanghai, People's Republic of China)

Bone-conduction stimuli evoked otoacoustic emissions could be clearly recorded in normal-hearing subjects. The sound probe was a 1/8-in. microphone mounted in a rubbery earplug inserted in the ear canal. A bone-conduct vibrator (RADIOEAR B-71, which was formerly used in an audiometer) as a stimulator, was applied to the surface of arcus of zygmaticus. Stimuli were generated by delivering 0.1-ms rectangular electrical pulses to the vibrator. With the exception of stimuli, the recording procedure was the same as Kemp's [J. Acoust. Soc. Am. 64, 1386–1391 (1978)]. By comparison in the same ear, both wave patterns and frequency spectra of the response evoked by bone conduction were quite similar to those evoked by click sounds at the stimulus levels approximate to the subjective thresholds. Because the frequency components above 800 Hz of skull vibrations go directly to the inner ear rather than being conducted through the middle ear bones, the finding of the present study provides new evidence to support the hypothesis that otoacoustic emissions originate from the cochlea. The new method of applying the stimulus also has the advantage of simplifying the structure of the detective probe and improving its frequency response.

10:15

C8. Reciprocal relation between the growth of an emitted cubic distortion product and the suppression of a spontaneous otoacoustic emission. E. G. Pasanen, C. C. Wier, and D. McFadden (Department of Psychology, University of Texas, Austin, TX 78712)

Emitted cubic distortion products were generated in ears having a spontaneous otoacoustic emission (SOAE). The frequencies of the primary tones were set to yield a distortion product 20 Hz below the frequency of the SOAE; the levels of the tones were equal and were varied over a wide range. With primaries below about 35 dB SPL, both emissions were present and could be measured separately. As the levels of the primaries were increased, and the amplitude of the distortion product rose, the SOAE was progressively suppressed from its level in the quiet until, at some level of the primaries, it was no longer measurable. Over the range where both emissions were detectable, the rate of decline of the SOAE as a function of the level of the primaries was approximately slope -1.0, while the rate of growth of the emitted distortion product was approximately slope $+1.0$—suggesting perfect energy conservation. For higher primary levels, where the SOAE was no longer measurable, the growth function showed saturation, being of lower slope, or even nonmonotonic in some cases. Moderate doses of aspirin abolished the SOAEs of all subjects, but only slightly diminished the distortion products. [Work supported by NINCDS Grant NS 15895.]

10:30

C9. Cochlear model for acoustic emissions. M. Furst and M. Lapid (Department of Electronic Systems, Faculty of Engineering, Tel-Aviv University, Ramat-Aviv 69978, Israel)

Acoustic emissions, both spontaneous and evoked, were measured in humans and in several animals. There were significant differences in the experimental results. In all animals a strong acoustic distortion product (ADP) was obtained, whereas in 60%–70% of the human subjects ADP was unmeasurable. Those human ears that emitted ADP usually possessed spontaneous emission (SE) and also Kemp's echo. The amplitude of the ADP was stronger when its frequency or the primary-tones frequency was in the vicinity of SE or near a peak of the Kemp's echo spectrum. In animals, the SE and Kemp's echo were rarely measurable. In order to understand the differences of the emission properties observed in various people and animals, a modification of the cochlear nonlinear transmission-line model is suggested. On each branch of the transmission line, a noise source was added to simulate cochlear internal noises. To simulate a local cochlear damage, the vertical resistance was modulated. The emitted spectrum predicted by the model contained a frequency whose CF is at the location of the damage. When introducing a two-tone stimulus (f_1 and f_2), the emitted spectrum predicted by the model contained components of the DP frequencies (i.e., $2f_1 - f_2$ and $2f_2 - f_1$). The amplitude of the DPs increased when there was a local damage at the CF of either the primary tones or the DP frequencies. Following this research, it is concluded that (a) normally ADP in humans is much smaller than in animals because of the different loads at the oval window, and (b) a local damage in the basilar membrane causes the rise of spontaneous emission and an increase of ADP amplitude.

10:45

C10. Statistical properties of auditory threshold selection in guinea pigs. Linda J. Hood, Elizabeth K. Barlow, and Charles I. Berlin (Kresge Hearing Research Laboratory, Department of Otorhinolaryngology, Louisiana State University Medical Center, New Orleans, LA 70112)

Measurement of electrophysiological thresholds for specific frequencies in animals has been limited due to spectral problems when using brief broadband pulse stimuli necessary to elicit synchronous neural discharge. Frequency specific stimuli usually fail to yield responses any closer than 10–20 dB to actual behavioral thresholds. When a compound action potential obtained from a pulse or tone-burst stimulus is subtracted from a potential obtained with the original stiumuls plus a low-intensity continuous pure-tone masker, estimates of thresholds are near behavioral thresholds than measurements obtained using short duration stimuli alone. The effects of presentation rate (27.7 to 77.7/s), number of sweeps (625 to 10 000), and level of the continuous tone stimulus (-10 to 50 dB SPL) in a series of guinea pigs have been examined. Optimal recording conditions yielded thresholds to sine waves that are within 5 dB of behavioral data presented by Heffner *et al.* [J. Acoust. Soc. Am. 49, 1888–1895 (1971)], and somewhat less sensitive than data presented by Prosen *et al.* [J. Acoust. Soc. Am. 63, 559–566 (1978)]. [Work supported by DRF.]

11:00

C11. Individual differences in auditory-evoked potentials: Variability of middle-latency responses, including comparisons with brain-stem AEPs. Judith L. Lauter and Roanne G. Karzon (Department of Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721 and Department of Otolaryngology, Washington University School of Medicine, St. Louis, MO 63110)

The variability of auditory-evoked potentials has been studied based on a repeated-measures design where individuals are tested under the same conditions in a series of eight weekly sessions. In previous reports to this Society, it has been shown that for auditory brain-stem responses (ABRs): (1) within-subject variability of waveform parameters such as peak latency and amplitude is smaller than between-subject variability; (2) individual subjects show characteristic variability profiles which can, in turn, classify individuals into groups; and (3) variability measures reveal differences in auditory-system response to right, left, and binaural stimulus presentations. In this paper, these measures are extended to middle-latency responses (MLRs), represented by three vertex-negative and two vertex-positive peaks. As expected, variability profiles for MLRs show much higher degrees of variability than for ABRs, even in within-individual comparisons; however, sensitivity to individual differences and ear of input are still apparent. In addition, our variability measure provides yet another means of distinguishing between the reputed brain-stem component of the MLR and later MLR peaks. [Work supported in part by AFOSR.]

11:15

C12. Effect of high stimulation rates and paired stimuli on the electrically evoked brain-stem response. D. D. Brown and R. T. Miyamoto (Department of Otolaryngology—Head and Neck Surgery, Indiana University School of Medicine, Indianapolis, IN 46223)

Temporal cues are a primary source of information for the single-channel cochlear implant subject. Two temporal features of the electrical auditory brain-stem response (EABR) in patients implanted with single-channel devices are being investigated. In the first of these studies the stimulus rate is varied from 10 to 150/s and changes in wave V amplitude and latency are recorded. Increasing the stimulus repetition rate causes a prolongation of wave V latency and a decrease in amplitude. These changes are particularly prominent for stimulation rates exceeding 70/s. In the second experiment the EABR response to the second stimulus of a stimulus pair is computed as the time interval between the two stimuli is varied from 0.5 to 10 ms. Initial data indicate a significant interaction only for interstimulus intervals less than 4 ms. In these two studies, variations across patients have been found and an attempt is being made to correlate these variations with other measures of the subjects' implant performance. Findings and clinical implications will be reported.

11:30

C13. Effect of acoustic tone perception on brain-generated electromagnetic waves: Preliminary evidence. Philip L. Stocklin (Mentec, Inc., 439 Blue Jay Lane, Satellite Beach, FL 32937), Glenn M. Cohen (Florida Institute of Technology, 150 W. University Boulevard, Melbourne, FL 32901), Norma L. Stocklin (Consultant, 439 Blue Jay Lane, Satellite Beach, FL 32937), and Ron G. Blackburn (Consultant, 3218 Wind Song Court, Melbourne, FL 32935)

Brain-generated microwaves were detected and measured for each of four subjects in the frequency range of the lowest electromagnetic standing wave (mode) of the adult human brain/skull cavity (200–400 MHz) in the presence and absence of a loud acoustic tone. In each case, the center frequency of the mode was found at, or within, a megahertz of that measured for each subject 1 yr earlier, and is different for each subject. In all cases, mode center frequency is within 10% of that theoretically predicted based on a subject's skull dimensions [P. L. Stocklin and B. F. Stocklin, T.I.T. J. Life Sci. **9**, 29–51 (1979)]. In the absence of the acoustic tone, mode structure consists of a cluster of frequency spikes, each 2 to 3 kHz wide extending over 30 to 50 kHz. In the presence of a loud 1500-Hz continuous acoustic tone, the cluster of narrow spikes coalesces to one or two narrow spikes, 3 to 10 dB above the cluster average. [Work supported by Mentec, Inc.]

11:45

C14. Signal detection analyses of neonates' responsiveness to sound. Lincoln Gray (Department of Otolaryngology, University of Texas Medical School, Houston, TX 77030)

Receiver operating characteristics (ROC curves) were generated from momentary delays in the regular peeping of newborn chicks that occur in response to acoustic stimuli. Areas under these curves have been used to evaluate the effects of age, intensity, and habituation. In the present paper, such ROC curves are used to evaluate the effect of prestimulus behavioral state on poststimulus responsiveness. Neonates are known to cycle through periods of high and low activity, and undoubtedly through periods of increasing and decreasing attention to external stimuli. Signal-detection analyses can be used to break the circularity in the argument that "neonates are more responsive when alert," and to help decrease the high variability in neonatal data. As expected, desirable prestimulus states are at moderate levels of activity; psychophysical performance is reduced during periods of high and low activity. Interestingly, the older subjects perform best when they are slightly more active than average, while the youngest subjects perform best when they are slightly less active than average. Similar analyses should be useful in other studies that depend on the changing rate of repetitive responses. [Work supported by NIH.]

TUESDAY AM

TUESDAY MORNING, 12 MAY 1987 — MT. McKINLEY, 8:30 A.M. TO 12:15 P.M.

Session D. Underwater Acoustics I: Propagation of Underwater Sound: Range-Dependent, Three-Dimensional, and Nonlinear Media

David F. Gordon, Chairman
Naval Ocean Systems Center, Code 711, San Diego, California 92152

Contributed Papers

8:30

D1. The finite element method applied to ocean acoustic propagation. Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148) and Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL, MS 39529-5004)

The finite element method offers the possibility of a full-wave solution for range-dependent acoustic propagation. It allows an accurate treatment of the basic physics of the problem in realistic range-dependent environments, and thus, for those problems not amenable to exact analytic solutions, it could serve as the benchmark against which adiabatic mode, coupled mode, parabolic equation, and other models might be compared. The two major problems to be solved in implementing a finite element model for ocean acoustic propagation are (i) the huge systems of equations which must be solved, and (ii) accurate treatment of the radiation condition at the computational boundary of the model. Solutions for both of these problems will be proposed.

8:45

D2. Critical test of Gaussian beam method for multiply reflected fields in a surface duct. E. Niver, C. J. Ruiz, M. S. Vogas (Department of Electrical Engineering, New Jersey Institute of Technology, Newark, NJ 07102), and L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, NY 11735)

The Gaussian beam method (GBM) has found wide application in seismology for charting high-frequency wave propagation through heterogeneous environments. Unlike ray fields, Gaussian beams negotiate convergence zones without requiring caustic corrections. Because of these attributes, GBM may be useful also for sound field propagation in a realistically range- and depth-dependent ocean. However, in its conventional and facile implementation via a discretized stack of paraxially approximated beams, GBM is not a discipline with *a priori* predictive capability. By comparison with various canonical test problems, the accuracy of GBM has been found to depend strongly on the choice of arbitrarily assignable beam and stacking parameters. Here, the method is tested on previously not investigated multiply reflected fields in a range-independent duct bounded by a rigid bottom. The fields on the boundary are computed by asymptotic ray theory (ART), GBM, and by a rigorous generalized ray integral whose numerical evaluation serves as a reference. It is found that a beam stack "tuned" for accuracy along a ray that has undergone a given number of reflections becomes successively less accurate as the observer moves to a ray segment with a lesser or greater number of reflections. Retuning removes this deficiency but requires again a reference result for comparison. This circumstance illustrates again the basic difficulty with GBM. Possible spectral remedies [Lu, Felsen, and Ruan, Geophys. J. R. Astron. Soc. (1987)] are discussed, but they are less readily computable. [Work supported by NJIT and ONR.]

9:00

D3. Rapid computation of acoustic fields in three-dimensional ocean environments. Michael B. Porter, W. A. Kuperman, F. Ingenito (Naval Research Laboratory, Washington, DC 20375), and Andrew A. Piacsek (Planning Systems, Incorporated, McLean, VA 22102)

A wave-theory-based technique for the rapid computation of propagation loss over a wide geographical area with a complex three-dimensional environment has been developed. The method relies on an adiabatic normal mode approach that reduces the wave equation to a set of coupled equations in the depth variable z and the plan variables (x,y). The coupled equations are solved with a fast algorithm based on dividing the (x,y) domain into triangular patches. A unique feature of the technique, which greatly speeds the computation, is the precalculation of an impedance surface that replaces the oceanographically stable portion of the sound-speed profile and the geo-acoustic structure of the ocean bottom, both derived from archival information. Once the modes are calculated, new source/receiver configurations are easily treated since the modes do not need to be recalculated. Because of the precalculated nature of the algorithm, areas with complex topography or bottom structures require no additional running time over simpler environments. As an example, a deep water scenario containing both a seamount and an eddy is considered.

9:15

D4. Low-frequency underwater acoustic waves from an ice plate-edge leaky Rayleigh wave. Jacques R. Chamuel (Sonoquest/Advanced Ultrasonics Research, P.O. Box 153, Wellesley Hills, MA 02181)

Leaky Rayleigh waves propagating along the bottom side of floating sea-ice plates experience cylindrical geometrical spreading and have their low-frequency limit determined by the ice plate thickness that corresponds to about 560 Hz. A second type of leaky Rayleigh wave can exist along ice plate edges propagating with no geometrical spreading and has its low-frequency limit determined by the large plate horizontal dimensions. Laboratory ultrasonic modeling results are presented demonstrating the detection of low-frequency Rayleigh waves radiated from ice plate edges. In one experiment, the plate-edge leaky Rayleigh wave wavelength was greater than 12 times the ice plate thickness. The results suggest that a 3-m-thick ice plate in the Arctic may easily radiate 10- to 100-Hz components from its edges. The ice plate edge acting nearly as a finite length line source produces high directionality of the radiated leaky Rayleigh wave into the water. The low-frequency plate-edge leaky Rayleigh waves may be detected in the water at significant distances from the ice edge. Additional physical insights into the controversial leaky Rayleigh wave are provided by comparing water/ice and Freon-113/ice interface conditions where a high-density low-velocity liquid is used. [Work supported by ONR.]

9:30

D5. Time domain propagation studies with the NPE: Two short movies. B. Edward McDonald (Naval Ocean R&D Activity, NSTL, MS 39529), Dan Plante, John Ambrosiano (Berkeley Research, Inc., Springfield, VA 22150), and William A. Kuperman (Naval Research Laboratory, Washington, DC 20375)

The nonlinear progressive wave equation (NPE) model has been used to investigate the propagation of linear and nonlinear broadband signals in a range-dependent shallow water waveguide. The NPE is the nonlinear time domain equivalent of the frequency domain parabolic wave equation (PE). Results from each of the studies are given in short computer-generated movies illustrating a four-cycle sine-wave pulse propagating a total of 20 km in isovelocity water ($c = 1500$ m/s) over a bottom ($c = 1600$ m/s) whose depth profile contains a shelf of slope 0.15 connecting regions of constant depth 225 and 150 m. Effects evident in the movies are: radiation into the bottom resulting in head waves that re-radiate back into the water column; energy dumping following reflection from the sloping bottom and surface; and mode separation at late times. Early time results suggest that bottom penetration is greater when nonlinearity is included.

9:45

D6. Comparing effects of different starting fields for parabolic equation models. E. Richard Robinson and David H. Wood (Code 3332, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

A parabolic equation model requires a starting field; that is, the values of the field must be given as a function of depth at a fixed range. A desirable starting field must satisfy two requirements: (1) It should excite the propagating modes with the proper amplitudes; and (2) it should excite nonpropagating modes only to the extent that their effects can be filtered out with increasing range by the false bottom usually added under the ocean floor, which can more than double the depths involved. In practice, the usual approach is to use an easily computed result such as a Gaussian, filtered Gaussian, sinc, or uniform ocean starting field. The false bottom is then increased to meet the second criterion. The advantage of the normal mode starting field is that it does not excite unwanted nonpropagating modes, which minimizes the need for the false bottom, which in turn reduces the running time of the model proportionally—once the starting field is computed.

10:00

D7. A shallow-water range-dependent acoustic propagation problem with a stepwise coupled mode solution. Richard B. Evans (ODSI Defense Systems, Inc., North Stonington Professional Center, North Stonington, CT 06359) and James M. Syck (Code 3331, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

A particular shallow-water range-dependent environment was chosen and the stepwise coupled mode model COUPLE was applied. The problem consisted of a refracting water layer overlying an isospeed liquid sediment. The water depth varied between 120 and 210 m over a range of 50 km. The calculations were done at 50 and 300 Hz. The main range-dependent effect considered was the redistribution of energy due to forward and backscatter caused by both large and small scale bathymetic changes. The large scale changes, that evolve slowly with range, produced a smooth transition of energy between different sized sets of locally propagating modes. The more rapid, small scale, changes caused a pronounced energy loss due to forward scattering into highly attenuated high-angle modes. The effects of backscatter were negligible. The results are compared in this paper to IFD/PE and, in a subsequent paper, with other parabolic equation models.

10:15

D8. A comparison of parabolic equation models with stepwise coupled modes in a shallow-water range-dependent environment. James M. Syck and Joseph M. Monti (Code 3331, Naval Underwater Systems Center, New London, CT 06320)

Parabolic equation calculations are compared with stepwise coupled mode COUPLE calculations for a set of runs for which the two models have overlapping capability. The PE calculations were then extended to consider the additional case of refracting sediment. Normal mode start-up fields were used in the calculations. PE models have capabilities differing in the areas of wide angle capability, treatment of the bottom, and numerical methods. The intent of this paper is to discuss which of these differences make a difference. The environment chosen was that presented in the previous paper. Calculations were performed at 50 and 300 Hz. Water depths varied somewhat, but averaged about 150 m, corresponding to 3 wavelengths and 15 wavelengths at 50 and 300 Hz, respectively. The dependence of depth with range took two forms. The first consisted of smoothly varying linear segments, the second was the same as the first with a number of spikelike irregularities added in. Differences are discussed in terms of numerical issues and the physics incorporated in the models.

10:30

D9. Hybrid ray-PE analysis of guided high-frequency propagation. T. Ishihara (Department of Electrical Engineering, National Defense Academy, Hashirimizu, Yokosuka, 239 Japan) and L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, NY 11735)

Surface guiding of high-frequency sound fields may take place near a concave boundary in a homogeneous medium or near a plane boundary in

a refracting medium. Multiple reflected ray fields excited by a source near the boundary form caustics that pile up at long ranges around the source depth. Conventional caustic corrections are inadequate to deal with this catastrophe. It is shown here that the narrow angular spectrum interval occupied by the troublesome caustic-forming rays can be filled alternatively by a narrow-angle parabolically approximated (PE) wavefield. The hybrid combination of legitimate rays and narrow-angle PE can be regarded either as correcting via PE the failure of ray acoustics at long ranges near the source depth, or as removing the narrow-angle restriction from PE by filling the wide-angle spectra with rays. The theory is illustrated with calculations for a range-independent ducting environment, and also for a range-dependent ducting-to-antiducting transition. [Work supported by ONR.]

10:45

D10. HYPER: A hybrid parabolic equation-ray model for underwater sound propagation in the vicinity of smooth caustics. Frederic Tappert (University of Miami/RSMAS, Miami, FL 33149) and Henry Weinberg (ODSI Defense Systems, Inc., North Stonington Professional Center, North Stonington, CT 06359-1738)

The hybrid parabolic equation-ray model (HYPER) uses modified ray tracing of geometrical acoustics to locate important propagation paths. Then HYPER solves a modified parabolic equation in a small region near the rays to compute the pressure field. This paper applies the HYPER model to investigate acoustic propagation in the vicinity of smooth caustics. Results are compared with ordinary ray theory, modified ray theory, and the wave solution for a range-independent case. Some range-dependent examples are also presented. [Funding for this work was provided by the AEAS Program Office, ONR Detachment, Code 132.]

11:00

D11. Fast field computations using transmutation. Robert P. Gilbert (Department of Mathematical Sciences, University of Delaware, Newark, DE 19716) and Daivd H. Wood (Code 3332, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

Fast field computations without range dependence are based on computing the Hankel transform of the Green's function of the separated depth-dependent ordinary differential equation. As usual, the required Green's function is expressed in terms of two solutions of the homogeneous differential equation, the first satisfying the boundary condition at the surface of the ocean, the second satisfying the boundary condition at the bottom of the ocean, and the Wronskian of these two functions. In our approach, one solution is given in terms of one transmutation that preserves the *surface* boundary condition, and the other is given in terms of a *different* transmutation that preserves the *bottom* boundary condition. Once the two different transmutation kernels are found, they are applied to trigometric functions that satisfy the respective boundary conditions, and the Green's function is numerically computed. A fast Hankel transform (the usual fast Fourier transform approximation was used) completes the calculation of the field versus range.

11:15

D12. Numerical solution for the diffraction of internal sound sources by a free-flooded cylindrical shroud. Rudolph Martinez (Cambridge Acoustical Associates, Inc., 54 CambridgePark Drive, Cambridge, MA 02140)

The problem of determining virtual source distributions for an insonified free-flooded pipe of finite length is posed as a pair of coupled integral equations. The geometry is assumed to have a rigid outer surface and a locally reacting inner surface. Predictions are made of directivity patterns of diffracted sound, with particular interest in the effect of lining characteristics on levels of on-axis noise. At each chosen frequency the solution to the boundary-value problem is checked by confirming the power balance between that injected at the incident source, and the sum of corresponding radiated and lining-dissipated values. The effect of finite length on input energy is established upon comparison to that for the similarly lined infinite waveguide.

11:30

D13. The airborne noise path for an acoustic source inside a fluid-loaded cylindrical shell. Edmond Y. Lo (Cambridge Acoustical Associates, Inc., 54 CambridgePark Drive, Cambridge, MA 02140)

The radiation of noise from machinery located inside a submerged elastic structure involves both a direct structure-borne and flanking airborne path. Calculations were performed to assess the importance of the latter for a cylindrical shell excited by an interior monopole source. An analytic solution for the interior pressure field is derived for a prescribed wall impedance using the Helmholtz integral equation approach. Farfield pressures are evaluated for a baffled section of the shell. Transfer functions of farfield to interior pressures are presented over a frequency range encompassing the ring frequency. The excess insertion loss arising from the interior application of various absorptive blankets is evaluated.

11:45

D14. Resolution of low-frequency attenuation anomalies in the Southern Hemisphere. R. N. Denham, R. W. Bannister, K. M. Guthrie (Defence Scientific Establishment, Naval Base Post Office, Auckland, New Zealand), R. H. Mellen (PSI—Marine Sciences, 95 Trumbull Street, New London, CT 06320), and D. G. Browning (Naval Underwater Systems Center, New London Laboratory, New London, CT 06320)

During Project KIWI ONE [D. G. Browning *et al.*, Nature **282**, 820–822 (1979)] anomalously high values of low-frequency attenuation were observed in the central South Pacific Ocean. Similar results were obtained along a track from New Zealand into the Southern Ocean during Project TASMAN TWO [R. W. Bannister *et al.*, J. Acoust. Soc. Am. **62**, 847–859 (1977)]. A recent analysis of oceanographic data by Mellen shows that both regions are included in a relatively high *p*H contour at the sound channel axis. The corresponding predicted values of attenuation are in reasonable agreement with the measured data.

12:00

D15. Anisotropy of compressional velocity in sea ice cores. Paul J. Vidmar and Jo B. Lindberg[a] (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

Laboratory measurements of the propagation velocity of ultrasonic compressional wave pulses were made at four orientations (vertical and at three horizontal directions) on five cores from first year sea ice. Accurately measured relative velocities ($\pm 0.9\%$) showed that velocities were slightly anisotropic (generally $<5\%$). As in freshwater ice, it was found that the velocities parallel to the *c* axis (which was in the horizontal plane) are faster than horizontal velocities perpendicular to the *c* axis. However, the measured vertical velocity is faster than velocity parallel to the *c* axis in our sea ice cores. This is surprising since the velocity along the *c* axis in freshwater ice is faster than the perpendicular velocity. Velocities at 45° to the *c* axis were found to be either faster or slower than velocities parallel to the *c* axis, depending primarily on temperature. [This work was supported by the ONR.] [a] Currently at BDM Management Services Co., Austin, TX 78759.

Session E. Architectural Acoustics I: Research in Building and Room Acoustics

Steven M. Brown, Chairman
Steelcase Inc., TC-1W-04, P.O. Box 1967, Grand Rapids, Michigan 49501

Chairman's Introduction—9:00

Contributed Papers

9:05

E1. Comparison of a fan-shaped and a rectangular hall. J. S. Bradley (Institute for Research in Construction, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

Newer auditorium acoustics measures were used to evaluate and compare a fan-shaped and a rectangular auditorium. Measurements included reverberation times, early decay times, clarity C80, overall strength, and lateral fractions each measured in octave bands from 125 to 8000 Hz. Measurements were made at combinations of 3–5 source positions and 12 receiver positions. The results indicate that the shape of the hall influences the within-hall variation of these acoustical quantities. The fan-shaped hall is thought to have a less diffuse early sound field that causes larger seat-to-seat variations in acoustical quality and a systematic variation in acoustical characteristics from the front to the rear of the hall.

9:20

E2. Theoretical functions of transient diffusibility in reverberant rooms. Yu-an Rao (Institute of Physiology, Academy of Sciences, Shanghai, People's Republic of China and Department of Psychology, University of California, Los Angeles, CA 90024) and Edward C. Carterette (Department of Psychology, University of California, Los Angeles, CA 90024)

Much effort has been spent in investigating the relationship between the physical properties of reverberant sound fields and the parameters of sound qualities. A single form of parameter, which is a function of both space and time, that might help in establishing a relationship between its real-time value and subjective criteria is sought. Based on a simplified model of spatially directional diffusibility, a general solution to the differential equation of transient diffusibility $d(t)$ and the definiteness ratio $D(t)$ is given. Computed and measured curves are consistent with each other; in particular, an explanation of the nonmonotonic V shape of $d(t)$ curves by means of theoretical calculations and mathematical derivations is provided. It is suggested that $d(t)$ and $D(t)$ could serve as quantitative, predictive parameters in applications to architectural acoustics and the control of sound quality in acoustic engineering.

9:35

E3. Measurement of speech privacy in open plan offices. J. B. Moreland (Westinghouse Electric Corporation, Research and Development Center, 1310 Beulah Road, Pittsburgh, PA 15235)

The proper design of an open plan office for speech privacy and for overall acoustical comfort considers many variables, all of which interact to produce the final acoustical environment. Using a loudspeaker to represent the talker, the sound spectrum that would be heard by a listener in an adjacent workstation is measured. From these measurements, the articulation index (AI) between adjacent workstations is calculated. Adjacent workstations are focused upon because it is generally more difficult to obtain good speech privacy for the occupants of these workstations than it is for workstations that are farther apart. Also, a PNC35 spectrum is used as the baseline masking spectrum. The PNC35 has an overall level of 43 dBA, and is representative of low-to-moderate levels of sound masking. A few trends emerge from this study. The sitting of workstations near hard room boundaries can be expected to have poorer speech privacy than corresponding workstations away from such room boundaries. Second, the acoustically hard screens whose NRC is 0.1 or less will have poorer speech privacy than will acoustically soft screens whose NRC is 0.5 or more.

9:50

E4. Study of acoustic response of anisotropic steel structures. P. Cordonnier, D. Biron, and S. Pauzin (ONERA/CERT/DERMES, 2 Avenue Edouard Belin-B.P. 4025, 31055 Toulouse Cedex, France)

The purpose of our experiment is to learn the influence of stiffening on the acoustic response of steel plates. The investigation is carried out on a rectangular plate that is clamped in an opening (1.20×2 m^2) between a reverberant room and a test room. The sound transmission loss is determined by using two different methods: a method that is based on the sound pressure measurement and the two-microphone intensity method. The mean radiation efficiency is obtained by measuring the panel vibration velocity with a regular mesh. The measurements are made over a frequency range from 250 to 10 000 Hz for a diffuse field excitation. The following series of samples is tested: a 1.5-mm-thick-steel plate; a plate reinforced with uniformly spaced tubular stiffeners; a plate reinforced with uniformly spaced line stiffeners; and a corrugated panel (the space between the corrugations is the same as the space between the stiffeners). The acoustic effect of variation of bending stiffness is discussed, and all the experimental results provide comparison elements for computed results.

10:05

E5. Sound transmission loss single number ratings. Keith W. Walker (USG Corporation, P.O. Box 460, Round Lake, IL 60073)

Several single number ratings, including STC, for rating the sound insulation properties of walls were correlated with the A-weighted sound reductions of a group of gypsum board partitions for several types of sound source. None of the ratings fared well for the broad range of sound types. It is shown that a single rating method cannot deal with all sound sources. Two new ratings are proposed to supplement STC for coverage of indoor and outdoor noise reduction.

10:20

E6. A source spectrum for simplified sound isolation tests specifically for results correlated to NIC ratings. Nigel D. Breitz (Acoustical Consultant, 212 Beit Megurim, Rehov Moshe Dayan, Kfar Saba, Israel) and H. G. Leventhall (Atkins Research and Development, Epsom, Surrey KT1 8SBN, United Kingdom)

Using one-third octave-band sound transmission and reverberation time data of 54 party walls (ISO717, ISO140), the differences in dBA values between rooms were simulated as a function of wideband (90 Hz–3.5 kHz) source room spectra. An optimal source room spectrum was obtained by minimizing the standard deviation of ($\Delta dBA_i - NIC_i$) ($i = 1.54$) from the mean value. The rank correlation coefficient was seen to increase for decreasing standard deviation. The absorption data of the rooms were used to extend the results to the sound power spectrum of the source. Results, examples, statistical data, and comparison with other workers will be given.

10:35

E7. On corner positioning of the sound source for sound isolation tests. Nigel D. Breitz (Acoustical Consultant, 212 Beit Megurim, Rehov Moshe Dayan, Kfar Saba, Israel) and H. G. Leventhall (Atkins Research and Development, Epsom, Surrey KT1 85BN, United Kingdom)

For reproducibility of simplified sound isolation tests [when a large ($\sim\frac{1}{2}$ m) approximately spherical, omnidirectional sound source is not used], it is appropriate to specify requirements for source placement. A scale model of a building with an exaggerated lopsided but reasonable quantity of absorption in one room (panel absorbers on wall surfaces around one corner) was used to investigate deviation of radiated power for the corner position from rigid wall conditions, the objective being to excite modes to an extent depending on their frequency and not mode number. The model and experiment will be discussed, and the results for this model as well as those for a full-sized baffled loudspeaker will be given. The placement is of more importance in countries where the 100-Hz, one-third octave band is included. In light of annoyance from hi-fi bass, the European range would seem more appropriate.

TUESDAY AM

10:50–11:00
Discussion

TUESDAY MORNING, 12 MAY 1987 — REGENCY BALLROOM D, 9:00 TO 11:30 A.M.

Session F. Physical Acoustics I: Scattering I

Philip L. Marston, Chairman
Department of Physics, Washington State University, Pullman, Washington 99164

Contributed Papers

9:00

F1. A rigid background analysis of the acoustic scattering from solid elastic targets. D. H. Trivett, Gary S. Sammelmann, and Roger H. Hackman (Naval Coastal Systems Center, Panama City, FL 32407)

At previous meetings of the Acoustical Society of America [L. H. Green, D. H. Trivett, and L. Flax, J. Acoust. Soc. Am. Suppl. 1 77, S79 (1985)], it was demonstrated that, for the analysis of the elastic excitations in the acoustic scattering from elastic targets, the proper background to subtract is a rigid background. This choice was utilized to analyze the low-frequency scattering from spherical and infinite cylindrical shells over a broad range of frequencies and shell thicknesses. However, in a recent paper on the acoustic scattering from elastic spheroidal solids [M. F. Werby and G. J. Tango, J. Acoust. Soc. Am. **79**, 1260–1268 (1986)], it is concluded that a rigid background is inappropriate for elastic targets that have either low shear speeds or low densities. A complete reexamination of the scattering from elastic solids is presented to demonstrate that the rigid background is the appropriate choice in this case, too. In the course of this analysis, it is also demonstrated that Werby and Tango have misinterpreted the nature of the elastic response of the scatterer.

9:15

F2. Time-frequency analysis of acoustic reflection waves from elastic spheres. N. Yen, Louis R. Dragonette (Physical Acoustics Branch, Naval Research Laboratory, Washington, DC 20375-5000), and Angie Sarkissian (Sachs/Freeman Associates, Landover, MD 20785)

A processing algorithm based on the Wigner distribution function is used to analyze the acoustic reflection from a solid sphere in a joint time and frequency representation. Test cases for spheres constructed from various elastic materials were computed from the theoretical form functions in the frequency range for *ka* from 0.3 to 20. It is observed from such time-frequency displays that the behavior of the scattered wave from those penetrable scatterers can be expressed in several related wave groups with certain frequency component concentrations and distinct time delays. Such features were exploited to reveal their relationship with the shape of the sphere and the velocity of sound wave propagation through the structure. The time separation among different delayed wave groups provides the characteristic parameters of the scatterer as their destructive interference will form strong nulls corresponding to the sharp spectrum lines predicted from the resonance scattering theory. Utilization of time-frequency analysis as a practical approach to describe a reflective elastic sphere is discussed.

9:30

F3. Classification of resonances and remote aspect-ratio determination of submerged spheroidal elastic shells. G. C. Gaunaurd (Naval Surface Weapons Center, White Oak, R-43, Silver Spring, MD 20903-5000) and M. F. Werby (Naval Ocean R&D Activity, Code 221, NSTL Station, MS 39529)

Acoustic scattering was studied from submerged, steel, spheroidal shells in their resonance region. Arbitrary angular incidences were considered to analyze the possible aspect-angle dependence of the resonances. The shells are sufficiently thin so that the "proper backgrounds" of the RST [G. C. Gaunaurd and M. F. Werby, Int. J. Solids Structures **22**, 1149–1159 (1986)] can be suitably suppressed, and the resonances can be adequately isolated. It is shown that the shell resonances are caused by surface standing waves propagating at discrete frequencies with an almost constant phase velocity. From this (surface) standing-wave interpretation (viz., geodesics) accurate resonance locations have been derived. Only two basic vibrational modes seem to exist, one along and the other normal to the axis of symmetry. Thus, only resonances corresponding to these two directions are possible. Furthermore, the fixed resonance locations seem to have amplitudes that change with incidence angle. Hence, this procedure permits the classification of resonances, and it can be exploited to extract the aspect ratio of the spheroidal shell from its returned echo. These points are illustrated with numerous computational examples. [Work supported by NAVELEX and ONR.]

9:45

F4. Acoustic resonance scattering by large aspect-ratio solid targets. L. H. Green, C. George Ku (Martin Marietta Baltimore Aerospace, Baltimore, MD 21220), and M. F. Werby (NORDA, NSTL, MS 39529)

Previously, Werby and Tango [J. Acoust. Soc. Am. 79, 1260–1268 (1986)] examined the resonance response of solid spheroidal targets as a function of material properties. Results were presented for several different materials for targets with an aspect ratio <2. Here the previous analysis is extended to high aspect-ratio targets. The transition matrix approach to acoustic scattering utilizing spherical basis states is used. Results for several different materials are presented over a $KL/2$ range of 0.5–16.0 for targets with aspect ratios from 1–10. It is demonstrated that the phase velocity of the Rayleigh resonances on solid spheriods is closely related to the shear speed of the material. In addition, high Q resonant peaks are shown to dominate the backscattered response for many high aspect-ratio targets.

10:00

F5. Scattering of acoustic waves by a finite cylindrical shell containing a mixture of wax and colophony immersed in water. X. Bao, V. V. Varadan, V. K. Varadan (Laboratory for Electromagnetic and Acoustic Research, Department of Engineering Science and Mechanics and Research Center for the Engineering of Electronic and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802), Q. Xu, and T. Wang (Institute of Acoustics, Academia Sinica, Beijing, People's Republic of China)

A method employing boundary and finite element approaches as well as model analysis is used to study the scattering of acoustic waves by a finite cylindrical shell immersed in water. The cylindrical shell contains a mixture of wax and colophony and thus the elastic wave propagation in this mixture is also considered. The computed frequency response and direction pattern of the backscattering target strength show excellent agreement with measurements.

10:15

F6. Coupling coefficient G for the Fabry–Perot representation of backscattering from spheres and cylinders: Quantitative GTD for elastic objects in water. Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

For each class of elastic surface wave, the contribution f_{sw} to the total form function may be written in a Fabry–Perot representation [K. L. Williams and P. L. Marston, J. Acoust. Soc. Am. 79, 1702–1708 (1986)]. For backscattering $|f_{sw}| = |G \exp[-2\beta(\pi - \theta_{sw})]/[j + \exp(-2\pi\beta + i2\pi xc/c_{sw})]|$, where $x = ka$ and for spheres $j = +1$. The surface wave's phase velocity and angular attenuation coefficients are c_{sw} and β while $\sin\theta_{sw} = c/c_{sw}$. The exact form function was synthesized by summing various f_{sw} with a specular term. An expression for G, which had to be evaluated by numerical differentiation, was derived for solid spheres via a Sommerfeld–Watson transformation. In the present research, a simple approximation for $|G|$ is derived by matching f_{sw} with the RST result near a resonance when β is small. This gives $|G| \approx 8\pi\beta c/c_{sw}$ for spheres while for circular cylinders $j = -1$ and $|G| \approx 8\pi\beta/(\pi x)^{1/2}$. Comparison with the exact $|G(x)|$ for Rayleigh waves on solid tungsten carbide and aluminum spheres gives agreement even when $2\pi\beta$ is not $\ll 1$. These approximations of $|G|$ should also apply to the scattering of short tone bursts and to the f_{sw} for hollow elastic shells. [Work supported by ONR.]

10:30

F7. Focused backscattering from hollow spherical shells in water: Lamb waves and the acoustical glory. Steven G. Kargl and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

When a short tone burst is incident on a spherical shell in water, the first backscattered echo is the specular reflection. This is followed by radiation from the Lamb waves on the shell [R. Hickling, J. Acoust. Soc. Am. 36, 1124–1137 (1964)]. The angular dependence of the largest Lamb wave echo has been modeled and measured as the receiver is shifted an angle γ away from the backscattering axis. The sphere was made of stainless steel with an inner-to-outer radius ratio $b/a = 0.84$. For the frequencies used, $50 < ka < 65$ and the largest Lamb echo is predicted to be the lowest flexural mode. Previous SWT results (for solid spheres) were applied to calculate the surface wave's phase velocity c_{sw} and angular damping parameter β and were extended to include a subtle effect of β on the Lamb echo amplitude p_{sw}. It is found that $p_{sw} \approx p_s A_0 |J_0(u) - i\beta\gamma J_1(u)|$, where p_s is the specular amplitude at $\gamma = 0$, $u \approx ka\gamma c/c_\omega$, $A_0 = |G|\exp[-2\beta(\pi - \theta_{sw})]$, $\sin\theta_{sw} = c/c_{sw}$, and G is a coupling coefficient. Measured p_{sw} near the central peak at $\gamma = 0$ generally agree, but near the γ where p_{sw} is minimized, dispersion (not included in the model) becomes significant. [Work supported by ONR.]

10:45

F8. Hyperbolic-umbilic focal sections: The wavefield and the merging of rays at caustic lines. Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

A wave whose amplitude in the xy plane is $\exp[ik(g - ct)]$ propagates to produce a hyperbolic-umbilic focal section (hufs) if $g(x,y) = (\alpha x^3 + 3\gamma y^2 x)/6 + a_1(x^2 + y^2)$ with $\mathrm{sgn}(\alpha) = \mathrm{sgn}(\gamma)$ and $a_1 < 0$. Such a wave may be produced, e.g., by reflecting sound from a curved surface. The hufs lies in a plane a distance $z = -(2a_1)^{-1}$ from the xy plane. Let u and v denote the transverse coordinates in the hufs such that the axes are parallel to the x and y axes, respectively. The uv plane contains caustic lines which meet at $(u,v) = (0,0)$ with an apex angle Ψ given by $\tan(\Psi/2) = (\gamma/\alpha)^{1/2} = \beta$ [P. L. Marston, J. Acoust. Soc. Am. Suppl. 1 80, S73 (1986)]. The present research describes how the rays merge so that the caustics locate a transition from a four-ray region to a uv region containing no rays. The diffraction integral was also approximated for the hufs wavefield in the case $\alpha > 0$ by changing g in an analysis previously given [P. L. Marston, J. Acoust. Soc. Am. 81, 226–232 (1987)] for the transverse cusp. The amplitude in the hufs is $(k^{1/3}2^{4/3}\pi/ir\alpha^{2/3}\beta)\exp(ikr)\mathrm{Ai}(q_1)\mathrm{Ai}(q_2)$, where $r = (z^2 + u^2 + v^2)^{1/2}$ and the Ai are Airy functions of the indicated arguments: $q_j = -k^{2/3}(u \pm \beta^{-1}v)/(2\alpha)^{1/3}z, j = 1,2$. [Work supported by ONR.]

11:00

F9. An iterative T matrix for scattering from bounded objects. Guy Norton, Richard Keiffer, and M. F. Werby (Numerical Modeling Code, NORDA, NSTL, MS 39529)

The extended boundary condition (EBC) method of Waterman has been used extensively over the last few years to describe scattering from bounded objects and usually leads to a mapping (the T matrix) that relates the known incident field with the scattered field. This method arises from an expansion of relevant physical quantities in a partial wave space (i.e., the incident and scattered fields) that upon truncation of the partial wave series leads to a finite matrix problem. The level of truncation relates to convergence and it is not always possible to know at what truncation level one has arrived at convergence. An iterative T matrix approach has been developed at NORDA that starts at some truncation level (N terms in the expansion) and leads via simple vector operations to the $N + 1$ case. This method can obviate the problem of convergence by interacting upward until enough terms are included to yield a stable solution. In addition to dealing with the convergence problem this method is able to treat the problem of ill-conditioned matrices often encountered in scattering problems more effectively than the direct matrix method. The derivation of this method is presented along with some numerical results.

11:15

F10. Acoustical holography based on Bojarski's exact inverse scattering theory. Woon S. Gan (Acoustical Services Pte Ltd., 29 Telok Ayer Street, Singapore 0104, Republic of Singapore)

This paper investigated the use of Bojarski's exact inverse scattering theory (BEIST) as the basis of acoustical holography (AH). The purpose is to improve generalized holography although generalized holography originates from BEIST. Generalized holography is not the complete solution of BEIST and some information is missing. Generalized holography (GH) has to be recast as an inverse scattering problem and not as an inverse source problem. By interpreting GH as inverse scattering, one takes account of sound diffraction and refraction within an inhomogeneous object, giving detailed information on the object that is not available in GH. In the present AH setup, more than one scattering experiment has to be performed using different insonifying waves of different frequencies to determine the object's properties. This is the same as what is done in diffraction tomography experiment. The holographic field equation was obtained using the BEIST. It included the scattering amplitude. A set of simultaneous equations was obtained relating the object (or scatterer) to the recorded scattering amplitudes as specified for some discrete sets of frequencies. These simultaneous equations were solved for the scatterer in terms of the measured scattering amplitudes. A new type of exact inverse scattering algorithm for AH is proposed.

Session G. Speech Communication II: Vowel Perception (Lecture and Poster Session)

Winifred Strange, Chairman
Department of Communicology, University of South Florida, Tampa, Florida 33620

Invited Papers

1:00

G1. Evolving theories of vowel perception. Winifred Strange (Department of Communicology, University of South Florida, Tampa, FL 33620)

In the past 15 years, research on the perception of spoken vowels has given rise to new conceptions of vowels as articulatory, acoustic, and perceptual events. This paper traces the evolution of vowel theory starting from "simple" target models in which vowels were characterized articulatorily as static vocal tract shapes and acoustically as points in an $F1/F2$ vowel space. Within this conceptual framework, two major problems in perception to be accounted for were the Speaker Normalization problem and the Target Undershoot problem. Theoretical developments have taken two directions: (1) "Elaborated" target theories have been hypothesized which account primarily for perceivers' solution of the Speaker Normalization problem; (2) dynamic specification models have been offered (a) to deal with the Target Undershoot problem and (b) to characterize the perceptual significance of vowel diphthongization. Findings of perceptual studies are summarized that motivate these theoretical developments and a characterization is offered of vowels as dynamic gestures giving rise to temporally distributed acoustic information that perceivers utilize in identifying coarticulated vowels. [Research supported by NINCDS.]

1:30

G2. Auditory-perceptual interpretation of the vowel. James D. Miller (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

An account of the simple vowels of American English will be given in terms of the author's auditory-perceptual theory. Data from many sources will be used to illustrate and support this view. In addition to theoretical suggestions concerning an auditory-perceptual transformation and a segmentation rule, the perceptualized spectral patterns associated with the nonretroflex vowels will be shown to fall into regions within a slab in a three-dimensional, auditory-perceptual space. These regions form a "vowel map" that by simple rotations can be related to various "vowel charts" such as those of Jones, Pike, or Fant. While the present theory is an elaboration of traditional formant-ratio theory, it will be shown that a concept of a perceptual reference is needed not only for talker normalization but also to disambiguate vowels not distinguished by simple formant-ratio theory. Additionally, loci for the retroflex and nasalized vowels illustrate the utility of describing vowel spectra by five numbers combined into three dimensions. Simple $F1$ by $F2$ descriptions are not adequate except for enigmatophiles. [Work supported by NINCDS and AFOSR.]

2:00

G3. Static, dynamic, and relational factors in vowel perception. Terrance M. Nearey (Department of Linguistics, University of Alberta, Edmonton T6G 2E7, Canada)

This paper reviews theories and empirical findings that bear on the perception of English vowels, with emphasis on the comparison of data-analytic pattern recognition studies with results from speech perception experiments. Methodological approaches and empirical results from research at the University of Alberta [e.g., T. M. Nearey and P. F. Assmann, J. Acoust. Soc. Am. **80**, 1297–1308 (1986)] will be compared and contrasted to work in other laboratories. Issues to be considered include: (1) the role $F1$ and $F2$ and that of fundamental frequency and higher formants; (2) "vowel-space and speaking rate normalization" (global speaker context); (3) vowel-inherent spectral change (diphthongization); and (4) local phonetic context (e.g., consonantal context). While all of the above factors have been shown to have reliable effects on vowel perception, the relative weight of such effects and the circumstances which may alter these weights remain far from clear. Analysis of these problems in current research and suggestions for future studies will be presented.

2:30–3:30

All posters will be displayed from 2:30 to 3:30 p.m. To allow all contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:30–3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00–3:30 p.m.

G4. Front vowel perception in fluent speech. Larry H. Small (Department of Communication Disorders, Bowling Green State University, Bowling Green, OH 43403)

This study investigated the perceptual effects of misarticulating the English front vowels /i,ɪ,e,ɛ,æ/ located in two-syllable target words. Subjects were instructed to shadow a taped prose passage containing the misarticulated target words. All misarticulations occurred along the height dimension, in reference to the traditional vowel quadrilateral. Also, all misarticulations involved the stressed vowel of each two-syllable target word. The subjects tended to repeat words containing the vowel misarticulations as heard on the tape, instead of restoring the target words to their original form, indicating that the articulatory changes were noticeable to the subjects. If the changes were not noticeable, a greater number of restoration responses would have occurred. Subjects appeared to rely more on the acoustic-phonetic properties of the misarticulated vowels than on the contextual constraints of the passage. Since vowels are more intense and greater in duration than consonants, it is possible that listeners utilized stressed vowel information as a perceptual anchor during word recognition. That is, stressed vowel information may guide a listener during word recognition. [Work supported by the Axe–Houghton Foundation, New York.]

G5. The influence of extraneous components on vowel $F1$ estimation. Brian Roberts and Brian C. J. Moore (Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England)

The contribution of extraneous components to vowel $F1$ estimation was investigated by constructing a continuum perceived as changing from /ɪ/ to /ɛ/ as $F1$ was varied. The phoneme boundary location is a measure of the perceived $F1$, and hence of the influence of the extraneous component. Adding an extraneous component whose frequency is an integral multiple of $F0/2$ gives a complex consisting only of components harmonically related to $F0/2$. This component may thereby be incorporated by acting as a "false harmonic." Adding a false harmonic on the lower skirt of $F1$, at a level above the vowel spectral envelope, shifts the phoneme boundary upwards. However, adding a component at any other frequency in this region has a similar effect. The effect of further adding several false harmonics remote in frequency from $F1$, at levels equal to the spectral envelope of the vowel, is currently being investigated. Preliminary results suggest that adding these components increases the incorporation of the component added in the $F1$ region, but only if that component frequency is an integral multiple of $F0/2$. [Work supported by the SERC and the MRC, U.K.]

G6. Measurements of vowels in isolation and in sonorant context. Marios Fourakis and James D. Miller (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

In Miller's auditory-perceptual theory vowels are hypothesized to occupy nonoverlapping zones in a three-dimensional space defined by the relative positions of the first three spectral prominences of the short-term spectrum and by a reference related to the speaker's characteristics. In order to validate target zones drawn on the basis of the data available in the literature, we have recorded two male and two female speakers of American English producing the nine pure vowels of English in isolation and in sonorant context. The recordings were digitized at 20 000 Hz with 14-bit precision and analyzed using ILS. The formant and fundamental frequency information was extracted from the ILS analysis files. The data are now in the process of being smoothed in the log-frequency domain and are being transformed into paths in the three-dimensional auditory perceptual space. By examining not only the pure vowels but also vowels in sonorant context, the aim is to establish the coarticulatory effects on relative spectral prominence positions of preceding and following [l] and [w]. [Work supported by NINCDS and AFOSR.]

TUESDAY PM

G7. Judgments of coarticulated vowels are based on dynamic information. David R. Williams (Haskins Laboratories, New Haven, CT 06511-6695)

Due to coarticulation, formant frequencies for vowels such as [ʊ] and [ɪ] are lower between labial consonants than in isolated productions. When asked to categorize stimuli from three-formant or analogous three-tone [ʊ]–[ɪ] and [wʊw]–[wɪw] continua matched on formant midpoint frequencies, listeners place their category boundaries closer to [ʊ] for the [w w] stimuli. However, listeners asked to judge the relative pitch height of the same three-tone stimuli do not show a shift in category boundary locations [Williams *et al.*, J. Acoust. Soc. Am. Suppl. 1 **74**, S66 (1983)]. To determine whether or not this lack of a shift resulted because midpoint frequencies of the [w_w] stimuli were isolated perceptually, subjects judged either vowel class or pitch height of the same stimuli with reference to isolated [ʊ] and [ɪ]. While vowel categorization results were similar to those observed previously, pitch height boundaries now exhibited a shift opposite to that for the vowel judgments. Thus listeners apparently *integrated* across most (or all) of the duration of the [w w] stimuli when judging their pitch. These findings indicate that auditory mechanisms which isolate and interactively process successive time-varying and more stable regions of the speech signal cannot account for the perception of coarticulated vowels. [Work supported by NICHD Grant HD-01994.]

G8. Perceptual dimensions of alaryngeal vowels. Michael D. Trudeau and Robert Allen Fox (Speech & Hearing Science, The Ohio State University, 324 Derby Hall, Columbus, OH 43210-1372)

The present study uses multidimensional scaling to examine the perceptual dimensions underlying the perception of alaryngeal vowels and compares these dimensions with those commonly found in MDS studies utilizing normally voiced vowels. A set of 11 vowels ([iɪɛæaʌoʊʊuɔɚ] produced by a competent esophageal speaker represented the stimulus vowels. These vowels were compared (pairwise) on a nine-point similarity/dissimilarity scale by 20 listeners and the resulting perceptual distance data were analyzed using INDSCAL. Three perceptual dimensions were obtained (significantly correlated with $F2$, $F1$–$F0$, and $F3$, respectively), which accounted for 67% of the variance. These dimensions are similar to those obtained in studies using normally voiced vowels and suggest that alaryngeal voice quality does not disrupt the use of these common perceptual dimensions (which depend only upon vocal tract configurations). The nature of these perceptual dimensions will be discussed in terms of both acoustic (i.e., formants) and auditory (i.e., bark-transformed formants) measures.

G9. The perception of reduced vowel stimuli. Laurie F. Garrison and James R. Sawusch (Department of Psychology, Park Hall, SUNY at Buffalo, Amherst, NY 14260)

Most theories of vowel recognition employ the first two spectral peaks ($F1$ and $F2$) or some transform of these as perceptual dimensions for the vowel space. This experiment was designed to investigate how humans categorize vowels that do not contain the usual $F0$, $F1$, and $F2$. Subjects were presented with single formant vowels (270–2938 Hz; $F0$ of 125 or 215 Hz) and asked to identify them using 12 vowel labels. The data show high-frequency stimuli were identified as front vowels and low frequencies were identified as back vowels. The [i] category was used for both extreme high and low frequencies. The tense vowel labels [u], [o], [a], and [i] predominated. A second set of stimuli consisting of tone analog to $F1$ and $F2$ (no $F0$) was presented to subjects for identification. These stimuli were also consistently categorized. As with the single formant stimuli, subjects used the tense vowel labels most often. Results will be discussed with reference to models and the nature of prototypes in vowel recognition. [Work supported by NINCDS.]

G10. A comparison of two models of human vowel recognition. D. Lauren Dutton, James R. Sawusch, and Kam-Cheong Tsoi (Department of Psychology, Park Hall, SUNY at Buffalo, Buffalo, NY 14260)

Two models of human vowel recognition were simulated and their performances compared. The first was proposed by Syrdal and Gopal [J. Acoust. Soc. Am. 79, 1086–1100 (1986)] and modified for implementation. Five dimensions composed of critical band differences between spectral peaks (including $F0$) were computed for each input stimulus. Identification was based on the best match of the input to a set of prototypes. The second was a full octave filtering model simulating central auditory processing. Parallel filters combined all spectral peaks within three critical bands into a single peak. The match between each stimulus and the prototypes was then used as the basis for categorization. The prototypes used in each model were based on Peterson and Barney [J. Acoust. Soc. Am. 24, 175–184 (1952)]. The performance of the two models in identifying vowels will be described. In addition, performance of the models was compared to data of human subjects for tone analogs of vowel and single formant vowels. Relative strengths and weaknesses of the two models will be discussed. [Work supported by NINCDS.]

G11. Vowel identification: Are formants really necessary? Amir J. Jagharghi and Stephen A. Zahorian (Department of Electrical and Computer Engineering, Old Dominion University, Norfolk, VA 23508)

It has generally been assumed at least since the time of the comprehensive study by Peterson and Barney [J. Acoust. Soc. Am. 24, 175–184 (1952)] that the formant locations in vowel spectra are the most significant cues to vowel identity. In this experiment vowel spectra were represented by two methods: (A) by the locations of the first three formants, and (B) by the overall smoothed spectral shape in terms of a discrete cosine transform of the power spectra. Stimuli consisted of four repetitions of the widely separated vowels /u/, /i/, /a/, spoken by each of 12 female and 12 male speakers ($4 \cdot 24 \cdot 3 = 288$ stimuli total). For each of the two spectral encoding methods, A and B, the vowel data were projected to a three-dimensional space such that the vowel categories would be well separated and the vowels within each category well clustered [S. A. Zahorian and A. J. Jagharghi, J. Acoust. Soc. Am. Suppl. 1 79, S8 (1986)]. Significantly better clustering was obtained with method B, based on overall spectral shape, than for method A, based only on the first three formant frequencies. Since these results are not based on perceptual experiments, no direct conclusion can be drawn regarding the perceptual importance of spectral peaks versus overall spectral shape for human perception of vowels. However, the results do indicate that automatic machine identification of vowels can be improved by parameterizing the overall spectral shape rather than only the spectral peaks.

G12. Why is the formant frequency difference limen asymmetric? Hynek Hermansky (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

Flanagan [J. Acoust. Soc. Am. 27, 613–617 (1955)] observed asymmetries in formant frequency difference limen in the perception of some synthesized vowel-like sounds and attributed those asymmetries to the proximity of neighboring formants. This hypothesis about the origin of the asymmetries in the perception of differences in vowel formant position is tangible and is widely accepted. The perceptual experiments of Hermansky and Javkin [J. Acoust. Soc. Am. Suppl. 1 80, S18 (1986)] showed that the asymmetries in formant frequency difference limen can be reversed by small changes in fundamental frequency in the experiment. This observation questions the contribution of neighboring formants as a factor in the perception of small differences in formant position. Results of experiments in the perception of synthetic vowel-like sounds will be presented. They suggest that the formant distribution was not a major factor contributing to perceptual asymmetries in Flanagan's experiment. The role of formant structure in vowel perception is discussed.

G13. Perceptual learning of vowels in a neuromorphic system. Todd M. Morin and Howard C. Nusbaum (Department of Behavioral Sciences, The University of Chicago, 5848 S. University Avenue, Chicago, IL 60637)

Linear neuromorphic systems assume that stimuli are perceived as linear combinations of a set of underlying features represented as the eigenvectors of the stimulus space. Learning in this type of system is an autoassociative process that induces prototypes for each perceptual category. Anderson *et al.* [Psychol. Rev. 84, 413–451 (1977)] proposed that human listeners learn vowel categories through just this type of computational mechanism. To investigate this claim, a linear autoassociating network was trained on a set of prototypical American-English vowels. Performance of this network was examined for learning and classifying vowels produced by an average male talker. In addition, the effects of different auditory coding representations were compared on recognition performance for the male vowels. Since it is necessary for the underlying feature space to be linearly independent, the perceptual representation of the vowels can affect learning. Furthermore, the extent to which the underlying feature space learned from a set of vowels is shared between talkers is investigated. For this type of network to be a plausible model of vowel perception, it must be capable of perceptual constancy across talkers. Finally, the effects of different learning algorithms on the development of vowel categories in perceptual space were compared. The results of these studies and their implications for simple, linear network models of speech perception are discussed.

G14. Identification of "hybrid" vowels in sentence context. James J. Jenkins and Winifred Strange (Department of Psychology, University of South Florida, Tampa, FL 33620)

It has been demonstrated that listeners can identify the intended vowel in a CVC syllable even when the vowel nucleus has been attenuated to silence, leaving only onglides and offglides (Silent-Center syllables). Verbrugge and Rakerd [Lang. Speech 29, 39–57 (1986)] constructed "hybrid" Silent-Center syllables by cross-splicing onglides and offglides of citation-form CVC syllables spoken by a male and a female talker such that the formant trajectories were discontinuous. Identification of the intended vowel in hybrid syllables was no less accurate than vowel identification of the single-talker Silent-Center syllables. The present study replicates this research with syllables spoken in a carrier sentence "I say the word /dVd/ somemore," using ten American-English vowels. Hybrid Silent-Center stimuli were prepared by crossing the stimulus sentences so that the sentence started with one talker and switched to the other talker after the silent portion of the test stimulus. Silent-Center Control stimuli were also prepared for each talker. The findings suggest that dynamic syllable structure provides talker-independent information about an articulatory/acoustic event. [Research supported by NINCDS.]

G15. Explorations in speaker normalization. John J. Ohala and Yoko Hasegawa (Phonology Laboratory, Department of Linguistics, University of California, Berkeley, CA 94720)

Some preliminary efforts are reported at speaker normalization based on the assumption that listeners are tacitly aware (a) that they are listening to an instrument (the human vocal tract) which when uniform (unconstricted) should have resonances whose lowest frequencies are spaced according to the ratios 1:3:5:7, etc., (b) that the exact frequencies of these resonances depend on the length of the vocal tract, and (c) that what is relevant in resonances from a nonuniform tract is the ratio of these lowest resonances to their uniform value. There are probably several cues the listener could use to estimate the size of the speaker; for starters, we use the mean (long-term) $F4$. The reference resonances for $F1_r$, $F2_r$, $F3_r$, i.e., those from the "uniform" tract, are 1/7, 3/7, and 5/7, respectively, of mean $F4$. The normalization consists in converting measured formant frequencies, $F1_m$, etc., into the ratios of $\log F1_m/\log F1_r$. The results of applying this normalization to the entire voiced portion of vowels in a variety of CVC syllables is reported. [Work supported by a Sloan grant to Berkeley Cognitive Science Program.]

G16. Evaluation of a spectrum target prediction model in speech perception. Masato Akagi (ATR Auditory and Visual Perception Research Labs., Twin 21 MID Tower, 2-1-61 Shiromi, Higashi, Osaka, 540 Japan)

A mathematical model of the target prediction mechanism based on psychological experiments [Furui and Akagi, 12th ICA Toronto, A2-6 (July 1986)] is evaluated. In continuous speech, spectral undershoot is produced by coarticulation and the speech spectrum seldom meets its ideal target. It is assumed for the model that the undershoot is compensated by a prediction mechanism in the auditory system, realizing the target spectrum as an internal representation of speech spectrum. In this paper, three viewpoints are discussed comparing the results of the model with those of psychological experiments. (1) The model compensates for a transitional part of connected Japanese vowels and decreases the length of the transitional part that produces spurious vowel spectrum. (These spectra are actually unperceived under any speaking conditions.) (2) A perceptual critical point, where the following vowel in Japanese C–V syllable is initially perceived, corresponds well with its physically estimated point by the model. (3) The model is also applicable for compensating the transitional part of consonants in Japanese syllables and achieves the extraction of stable acoustic features for consonants. The results indicate that the model recovers phoneme characteristics neutralized by the coarticulation and corresponds well to the human hearing mechanism.

TUESDAY PM

Invited Discussants

3:30

Peter Ladefoged

Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, California 90024

3:45

Michael Studdert-Kennedy

Haskins Laboratory, 270 Crown Street, New Haven, Connecticut 06510

Open Discussion

4:00–4:30

Session H. Underwater Acoustics II: Scattering

Jacques R. Chamuel, Chairman
Sonoquest/Advanced Ultrasonics Research, P.O. Box 153, Wellesley Hills, Massachusetts 02181

Chairman's Introduction—1:15

Contributed Papers

1:20

H1. High-frequency reflection and scattering of sound by ellipsoidally embossed surfaces. R. J. Lucas[a] and V. Twersky (Mathematics Department, University of Illinois, Chicago, IL 60680)

Earlier results for coherent reflection and incoherent scattering by random distributions of relatively arbitrary bosses on rigid or free base planes [V. Twersky, J. Acoust. Soc. Am. 29, 209–225 (1957); 73, 85–94 (1983)] are applied to rigid and free ellipsoidal bosses with axes large compared to wavelength. The asymptotic procedures used originally for circular cylinders and spheres (1957) and subsequently for elliptic cylinders [J. E. Burke and V. Twersky, J. Acoust. Soc. Am. 40, 883–895 (1966)] are generalized to triaxial ellipsoids, and results for the corresponding bosses are obtained by the original image method. Illustrative curves exhibit the effects of boss shape and orientation on the coherent reflection coefficients and incoherent differential scattering cross sections per unit area. To facilitate comparison with available low-frequency curves, the same isotropic and anisotropic surfaces were considered as before [R. J. Lucas and V. Twersky, J. Acoust. Soc. Am. 78, 1838–1850 (1985)] and emphasize anisotropic forward (specular) and backscattered effects for applications to sea–air, sea–bottom, and sea–ice programs. [a] Visiting from Department of Mathematical Sciences, Loyola University, Chicago, IL.

1:35

H2. Ray scattering from a collection of ice keels. David F. Gordon (Naval Ocean Systems Center, Code 711, San Diego, CA 92152–5000)

An ice-keel program that was described at the Spring 1985 meeting [J. Acoust. Soc. Am. Suppl. 1 77, S56 (1985)] is used to investigate the acoustic scattering properties of ice keels at high frequency. Keels are modeled as parabolas. Same sets of keels are generated using published distributions of spacing and sizes [O. I. Diachok, J. Acoust. Soc. Am. 59, 1110–1120 (1976)]. A large number of rays are directed toward the keel set and the emerging rays are collected into angular bins. Reflections from and refraction through the ice are employed. Specular reflection coefficients computed in this manner are similar to Diachok's published results above 3° grazing angle. No significant specular reflection is found at lower grazing angles. The scattered rays tend to be directed predominantly into angles near the specular angle. [Work supported by NOSC Independent Research.]

1:50

H3. Quasidipole and quasiquadrupole scattering from Arctic leads. Herman Medwin and Ken J. Reitzel[a] (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

The leading and trailing edges of a plane ice plate and the perimeter of a polynya represent arrangements of wedge scatterers that are mirrored in the adjacent, pressure-release, water–air interface. As a result, an edge that is exposed to low-frequency underwater sound will backscatter energy in a pattern that closely resembles radiation from an acoustic dipole. Likewise, the reradiation from a lead whose separation is small compared to a wavelength can resemble radiation from a quadrupole. The reason for this behavior, and its significance in the total low-frequency scatter from the Arctic canopy, is discussed in terms of wedge theory and the laboratory scale model experiments, which confirm this interpretation. [Work supported by ONR.] [a] Ocean Acoustics Associates, 4021 Sunridge Road, Pebble Beach, CA 93953.

2:05

H4. Dependence of low-frequency backscatter on the size and orientation of lineal Arctic scatterers. Michael J. Browne and Herman Medwin (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

The backscatter of low-frequency underwater sound from the Arctic ice canopy is generally caused by leads and ridges of a vast variety of sizes, orientations, and spacings. Laboratory scale-model experiments, guided by the theory of sound diffraction from wedges, have been used to determine the dependence of low-frequency backscatter on the geometrical parameters. The specific variation with the orientation and length of the lineal scatterer is demonstrated, and it is shown that the predominant scatter occurs when there is a section of the lead or ridge that is azimuthally perpendicular to the sound path. [Work supported by ONR.]

2:20

H5. The validity of the perturbation approximation for rough surface scattering. Eric I. Thorsos and Darrell R. Jackson (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195)

The accuracy of the first- and second-order perturbation theory approximations for randomly rough, pressure-release surfaces has been examined through comparison with exact numerical results based on solving an integral equation. A Gaussian roughness spectrum was chosen for the surfaces, and a Monte Carlo procedure was used to obtain the average bistatic scattering cross section. Results will be presented to show that the condition $h/\lambda \ll 1$, where h is the rms waveheight and λ is the acoustic wavelength, is not sufficient to ensure the accuracy of first-order perturbation theory. An additional constraint must also be imposed on the surface correlation length. [Work supported by ONR.]

2:35

H6. First-order three-dimensional scattering from rough interfaces and the effects of shadowing, local reflection coefficients, and surface gradient terms. Richard Keiffer and M. F. Werby (Numerical Modeling Division, NORDA, NSTL, MS 39529)

It is usual in the development of rough interface models to limit theory to two-dimensional models or to ignore shadowing and penetration into the surface as well as surface gradient terms that arise in the formal derivation of the equations. The second-order term has recently been examined to determine its importance in scattering from rough interfaces. The purpose of this study is to examine the effects of (a) shadowing, (b) penetration, and (c) surface gradient terms on the overall calculation of some representative surfaces. Moveover, a full three-dimensional model can allow for out-of-plane scatter, and this effect will also be examined. The method of treatment of penetration into the surface is via a local reflection coefficient. Preliminary results show that the majority of these effects are most important in backscatter, where they tend to enhance the magnitude relative to the forward direction.

2:50

H7. Numerical investigations of rough surface scattering. David H. Berman and John S. Perkins (Code 5160, Naval Research Laboratory, Washington, DC 20375)

An improved point matching method [H. Ikuno and K. Yasuura, Trans. IEEE Antenna Propagat. **AP-21**, 657–662 (1973)] is used to calculate surface scattering amplitudes for periodic irregular interfaces. Results of this method can be examined both for energy conservation and least-square satisfaction of boundary conditions. Because of the speed of this algorithm and simplicity of programming, it has been possible to examine fluid–fluid interface scattering in two dimensions, for grazing angles as low as 5 deg, and Dirichlet scattering for simple two-dimensional surfaces. Large-period surfaces with arbitrary structure within a period have been used to mimic random rough interfaces. Reasonable averaged scattering cross sections for the random rough surface problem have been obtained, using ensembles of 10 or 20 surface realizations. When the rough surface is described by a steep power-law spectrum, ensemble averaged cross sections may be dominated by a single ensemble member, in effect giving a false alarm. Doppler effects for moving surfaces have also been calculated.

3:05

H8. Modeling the general pdf of intensity in terms of the scattering parameters for WPRM. T. E. Ewart (Applied Physics Laboratory and School of Oceanography, University of Washington, Seattle, WA 98105)

The intensity correlations for WPRM can be predicted for forward scatter from solutions of the parabolic 4th moment when the space-time autocorrelation function, the scattering stength parameter γ, and the scaled range X are known. Thus, the intensity variance and its time/depth correlations can be predicted. However, the intensity pdf has not been predicted for all ranges of γ and X. T. Ewart and D. Percival [J. Acoust. Soc. Am. **80**, 1745–1753 (1986)] have shown that the generalized gamma distribution function (GGDF) models the probability distribution of intensity in forward scattering over wide ranges in γ and X. In this study the two parameters of the GGDF are obtained as functions of γ and X by fitting the distribution to many realizations of simulated intensity data. Realizations of the intensity were generated by Monte-Carlo techniques [C. Macaskill and T. Ewart, IMA J. Appl. Math. **33**, 1–15 (1984)]. The mapping of the distribution parameters to γ and X provides a convenient way to examine the behavior of the distributions as they vary smoothly from lognormal to exponential with increasing X for constant values of γ. A heuristic model of the mapping is presented. Implications and future expectations are discussed. [Work supported by ONR.]

3:20

H9. Extended time and space results from MATE. Terry E. Ewart and S. A. Reynolds (Applied Physics Laboratory and School of Oceanography, University of Washington, Seattle, WA 98105)

In the Mid-Ocean Acoustic Transmission Experiment (MATE) pulsed tones and Taylor-weighted FM slides at 2.083, 4.167, 8.333, and 12.5 kHz were received, after propagation over an 18.1-km wholly refracted Fermat path, at four tower-mounted transducers located at the corners of a 3-m vertical by 150-m horizontal rectangle oriented transverse to the raypath. Extensive oceanographic measurements are available to characterize the deterministic and stochastic sound velocity field of the scattering volume. The scattering parameters γ and X range over 145–31 500 and 0.094–0.016, respectively. Earlier publications focused on a 6-day intensity time series. The newly processed data extend the time series to 15 days and provide 8 days of data measured simultaneously at the horizontally separated receivers. These measurements of the complex field, together with the space/time sound velocity measurements, provide a unique data set with which to test predictions of the various acoustic field moments. Results will be presented for single receivers and vertically and horizontally spaced pairs. Theoretical predictions, where available, will be compared. Implications of the results to our understanding of WPRM will be discussed. [Work supported by ONR.]

TUESDAY PM

3:35

H10. Geometrical theory of diffraction for high-frequency coherence functions in a weakly random medium with inhomogeneous background profile. R. Mazar and L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, NY 11735)

The localization of high-frequency wave propagation around ray trajectories and the reflection, refraction, and/or diffraction of these local plane wave fields by boundaries, inhomogeneities, and/or scattering centers have been combined via the geometrical theory of diffraction (GTD) into one of the most effective means for analyzing high-frequency wave phenomena in complex deterministic environments. These constructs are here incorporated into a stochastic propagation and diffraction theory for statistical moments of the high-frequency field in a weakly fluctuating medium with inhomogeneous background profile, provided that the correlation length l_n of the fluctuations is small, compared with the scale of variation, but large, compared with the local wavelength $\lambda = 2\pi/k = 2\pi c/\omega$ in the fluctuation-free background, with k being the local wavenumber, c the local wave speed, and ω the radian frequency. The major analytical building blocks for coherence functions and paired random functions (PRF) include propagators described in local coordinates centered on the curved GTD ray trajectories in the deterministic inhomogeneous background environment; multiscale expansions in these coordinates to solve for statistical measures of the parabolically formulated ray fields; Kirchhoff or physical optics (PO) approximations for fields reflected from extended smooth surfaces; and "point scatterer" solutions for small scatterers and edges. The PRFs are useful for correlating incident and backward fields traversing the same propagation volume. The theory is illustrated for forward propagation in a fluctuating medium with inhomogeneous and caustic forming background, for reflections and refraction due to a plane or smoothly curved interface in such a medium, and for diffraction due to a wedge and a small scatterer. [Work supported by RADC and ONR.]

3:50

H11. An approximation for propagating the fourth moment through a fluctuating ocean. David H. Berman and Dalcio K. Dacol (Code 5160, Naval Research Laboratory, Washington, DC 20375)

In this paper an approximation for fourth moment and second moment (two-frequency) path integrals is presented. The approximation is similar in spirit and form to an approximation recently developed by Uscinski *et al.* [B. J. Uscinski, C. Macaskill, and M. Spivak, J. Sound Vib. **106**, 509–528 (1986)]. However, the present approximation has the advantage that, for the cross-frequency intensity, it factorizes to products of the cross-frequency coherence for long ranges when the latter is calculated in the same approximation. In addition, it is easily demonstrated that the present approximation is the second of a sequence of approximations that will converge to the exact path integral. The idea of the approximation is to replace paths appearing in phase structure functions by "classical" paths plus a number times the Green's function for the spreading or

Jacobi equation. The calculation then reduces to a phase screen calculation, with the coefficient of the Green's function appearing as the position at which a ray crosses the screen.

4:05

H12. The influence of channel boundaries on remotely sensed target resonances. G. C. Gaunaurd (Naval Surface Weapons Center, White Oak, R-43, Silver Spring, MD 20903-5000) and M. McCarthy (National University of Ireland, Galway, Ireland)

Target resonances are unique characteristics of scatterers and serve to identify them unambiguously. In the vicinity of environmental boundaries, the way resonances are perceived by sensors is different than in the absence of boundaries. Thus, even though the resonances themselves do not change, the way in which they manifest themselves in the scattering cross section of scatterers as sensed at the receiver, changes near an environmental boundary. A formalism is developed to study this change and to assess the effect of boundaries upon the way elastic scatterers are excited into resonant vibration by incident sound waves. This formalism reduces to an earlier one developed [J. Acoust. Soc. Am. **73**, 1–12 (1983)] for an infinite medium lacking boundaries. In general, the effect of boundaries on the resonance features present in the sonar cross section depends on the azimuthal wavenumber m, even for spherical scatterers. It also depends on the depths H and h of the elastic scatterer and the receiver, beneath the free surface $z = 0$ of the half-space. The boundary's influence is greatest for $m = 0$, or $H \ll 1$, or both. The method relies on addition and translation theorems for various wavefunctions.

4:20

H13. Resonance frequencies and the phase matching of helicoidal surface waves on impenetrable spheroids. Barbara L. Merchant, Anton Nagl, and Herbert Überall (Department of Physics, Catholic University of America, Washington, DC 20064)

The existence of the (complex) resonance frequencies of acoustic scatterers is explained by the generation of surface waves that match phases after repeated circumnavigations of the objects. The resonance frequencies can be predicted [B. L. Merchant *et al.*, J. Acoust. Soc. Am. **80**, 1754 (1986)] for prolate spheroidal objects, in the special case of axial incidence. Here sound-soft, prolate-spheroidal targets subject to *obliquely* incident signals are considered. The generated surface waves propagate along geodesics of helicoidal type, for which we obtain the condition for closing, and the set of discrete "pitch angles" at which closing takes place. An integral condition is formulated for the phase matching of helicoidal surface waves, using local wavelengths of Franz's surface waves. It is solved numerically for the complex resonance frequencies, found to agree closely with the $m > 0$ (i.e., containing azimuthal components) resonance frequencies obtained from an independent T-matrix calculation. This agreement confirms the validity of the principle of phase matching for the general case, and the accuracy of the T-matrix results. [Work supported in part by the Office of Naval Research.]

4:35

H14. Bistatic resonant scattering from elastic spheroidal shells and the resonance order. Michael F. Werby (NORDA, NSTL, MS 39529), Herbert Überall, and Anton Nagl (Department of Physics, Catholic University of America, Washington, DC 20064)

Bistatic scattering at a resonance frequency can determine the order of the resonance, as demonstrated experimentally [G. Maze and J. Ripoche, J. Acoust. Soc. Am. **73**, 41 (1983)] and theoretically [M. F. Werby and H. Überall, J. Acoust. Soc. Am., in press] for elastic bodies of separable geometry (cylinders, spheres). For obstacles of more general shape, mode mixing takes place, but it was found (e.g., for solid prolate spheroids) [H. Überall, M. F. Werby, S. H. Brown, and J. W. Dickey, J. Acoust. Soc. Am. Suppl. 1 **80**, S128 (1986)] that, for moderate aspect ratios (up to 3:1), one mode remains dominant in the bistatic pattern, and may serve to characterize the resonance, even while a crossing of Rayleigh and Whispering Gallery resonances takes place. This method is applied to the resonances of spherical and spheroidal shells, characterized by low-frequency resonances related to the plate Lamb-wave modes. Our application of the bistatic method confirms these relations, showing again that mode-order assignments remain valid for moderate spheroids' eccentricities. [Work supported in part by the ONR.]

Session I. Noise II: Evaluation of Hearing Conservation Programs

Larry H. Royster, Chairman
Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, North Carolina 27695-7910

Invited Papers

1:30

I1. Audiometric data base analysis (ADBA): Opening comments. Larry H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910) and Julia Doswell Royster (Environmental Noise Consultants, Inc., P.O. Box 144, Cary, NC 27511-0144)

The primary goal of audiometric data base analysis (ADBA) procedures is the determination of the effectiveness of an existing hearing conservation program in terms of the program's ability to prevent significant on-the-job noise induced hearing loss. To be able to achieve this goal in the real world, it is necessary that one or more reliable indicators be defined that can be implemented practically within the management and personnel framework of existing industrial hearing conservation programs. The paper presents some history-related information regarding past efforts in developing ADBA procedures and the efforts of the S12.12 Working Group (Evaluation of Hearing Conservation Programs), and the objectives for this special session on ADBA procedures. Some considerations for the requirements of an ADBA procedure to be useful in indicating the level of effectiveness of a hearing conservation program will be reviewed, as well as factors that have been shown or are assumed to significantly influence audiometric data and thereby affect the outcome of applying different ADBA procedures.

1:45

I2. The need for and benefits of audiometric data base analysis procedures. Alice H. Suter (Industrial Audiology, 1501 Red Oak Drive, Silver Spring, MD 20910)

Some noise-exposed employees are losing their hearing despite the implementation of hearing conservation programs (HCPs). Employers do not know how to evaluate HCP effectiveness, and audiograms are often filed away after testing. Employers lack guidance from professionals. They also lack regulatory guidance, since OSHA's hearing conservation amendment has no provisions for evaluating HCP effectiveness. Compounding the confusion is OSHA's policy not to issue citations for average noise exposure levels between 90 and 100 dBA when "the results of audiometric testing indicate that any existing controls and hearing protectors are adequately protecting employees." OSHA gives no guidance, however, as to the assessment of adequate protection. Audiometric data base analysis (ADBA) procedures can yield valuable information on HCP effectiveness. They encourage accountability by management, consultants, employees, and compliance officers. They provide feedback to supervisors, motivating them to promote and enforce good HCPs in their departments. They provide feedback to employees, which can change or reinforce their behavior, as appropriate, toward hearing protector use. ADBA procedures provide the means for identifying and the stimulus for improving poor HCPs, and they provide a source of satisfaction and potential reward for effective ones.

2:15

I3. Predicting hearing loss for noise-exposed populations using ISO 1999. Daniel L. Johnson (Larson-Davis Laboratories, 280 South Main, Pleasant Grove, UT 84062)

The International Organization for Standardization (ISO) is publishing a standard, commonly called ISO 1999, entitled "Acoustics—Determination of occupational noise exposure and estimation of noise-induced hearing impairment." The predictive procedures of this standard will be discussed. Three different nonoccupational noise-exposed data bases will be used to illustrate the relation between noise exposure, aging, and the screening procedures used in selecting the nonoccupational noise-exposed data base. The author will provide background for limiting the resulting hearing threshold level associated with both age and noise (H') to less than the simple addition of age (H) and noise-induced permanent threshold shift (N) by the equation $H' = H + N - HN/120$. Finally, a procedure for using ISO 1999 to predict the effects of changing noise exposures (such as workers exposed to 95 dB for 10 years, then 85 dB for 30 years) will be presented.

2:45

I4. Predicting TTS for a noise-exposed population. W. Dixon Ward (University of Minnesota, Department of Communication Disorders and Otolaryngology, 2630 University Avenue SE, Minneapolis, MN 55414)

One source of test–retest variability in serial audiometry is the temporary threshold shift (TTS) caused by prior exposure to sound above 75 dB SPL. In industrial audiometry, TTS is always convolved with the effects of other sources of variability, such as earphone placement, headband tension, instrument calibration, internal physiological noise level, listener criterion, "learning," etc., as well as with the "true" permanent change in hearing threshold level that is generally the only statistic of interest. However, enough is known about TTS that relatively valid prediction can be made of the mean and standard deviation of the TTS to be expected at test frequency A in an ear with a pre-exposure hearing threshold level of B dB, measured C minutes after exposure to a noise having spectral characteristics D, level E, duration F, and temporal pattern G. Two propositions will be examined: (1) that this knowledge can be utilized in order to "correct" group audiometric data for TTS, i.e., that TTS can be deconvoluted from the total variance, and (2) that adequacy of use of hearing protection devices can be assessed by means of TTS.

3:15

I5. Results from applying various audiometric data base analysis procedures to data provided to the ANSI S12.12 Working Group. Julia Doswell Royster (Environmental Noise Consultants, P.O. Box 144, Cary, NC 27511-0144) and Larry H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910)

The ANSI S12.12 Working Group has received over 20 audiometric data bases contributed by U.S. and Canadian industries from their hearing conservation programs (HCPs). The data include at least four approximately annual audiograms for about 12 000 male employees and at least eight audiograms for about 3000 male employees. The authors applied various analyses to the data to identify statistics which differentiate the level of protectiveness of the HCPs represented. Some statistics are based on variability in the audiometric thresholds (percentages of employees showing defined hearing changes either for better or for worse). Other statistics are based on the rates of change in group mean audiometric thresholds over time, compared to expected age-effect changes for nonindustrial reference populations. Desirable ranges for the statistics are suggested based on determining the results for non-noise-exposed and low-noise-exposed populations.

3:45–4:15

Panel Discussion

Session J. Physical Acoustics II: Nonlinear, Loss and Bubble Phenomena

Mack A. Breazeale Chairman
Department of Physics, University of Tennessee, Knoxville, Tennessee 37996

Contributed Papers

1:30

J1. Third-order effects in the propagation of finite amplitude stress waves. Kun-tien Shu and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The propagation of a finite amplitude stress wave through a homogeneous, isotropic elastic solid is analyzed for cases of one-dimensional dilatation and shear deformation. The strain energy function is expanded in invariant form as a fourth-order polynomial in displacement gradients. The displacement equations of nonlinear dynamic elasticity are solved by perturbation techniques that yield a uniformly valid approximation. The result for a dilatational wave, which matches previous predictions, is essentially like a planar wave in a fluid, aside from redefinition of the coefficient of nonlinearity. The primary nonlinear effect for a dilatational wave is encountered at the second order. In contrast, the shear wave case, which was not fully analyzed in previous investigations, has no analog in inviscid fluids. The first-order shear displacement leads to a second-order dilatational displacement, which then influences the shear displacement at the third order. The overall effect of nonlinearity is to produce amplitude dispersion, as well as a small phase shift in the waveform. [Work supported by the NSF.]

1:45

J2. Diffraction and nonlinear distortion in sound beams as interacting wave phenomena. J. H. Ginsberg, H. C. Miao, and M. A. Foda (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Discrepancies between the nonlinear King integral [J. H. Ginsberg, J. Acoust. Soc. Am. 76, 1201–1214 (1984)] for a sound beam and experiment are encountered in the nearfield in high-frequency, high-intensity situations. It is shown that the nonlinear, as well as linear, signal at all locations may be represented as groups of quasiconical waves converging and diverging on the axis, as they propagate away from the transducer. Nonlinear distortion to the nearfield arises from interaction of different transverse wavenumber modes in each group, and also from interaction between the groups. The latter effect ceases to be significant in the farfield, whereas the self-distortion arising in either group grows. The nonlinear King integral is shown to be the limiting form for long ranges. The results for nearfield locations are extremely challenging to evaluate, and an approximation that can be used to evaluate waveforms in some situations is discussed. [Work supported by the ONR, Code 1125-UA.]

2:00

J3. Nonlinear effects in focused sound fields. Timothy S. Hart (Department of Electrical and Computer Engineering, The University of Texas at Austin, Austin, TX 78712-1063) and Mark F. Hamilton (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063)

Focused finite amplitude sound fields are investigated with numerical solutions of the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation. The numerical solution is based on the algorithm developed by Aanonsen *et al.* [J. Acoust. Soc. Am. 75, 749–768 (1984)], who used a Fourier series expansion of the sound pressure to reduce the KZK equation to a set of coupled parabolic equations. The basic algorithm has been modified by introducing a coordinate system that follows the convergent geometry of focused sound fields. In this way, more efficient numerical evaluation of the detailed field structure within the focal region is achieved. Arbitrary axisymmetric sources can be modeled. Here, circular sources having linear focusing gains of order 50 will be considered. The calculated time waveforms, propagation curves, and beam patterns illustrate clearly the combined effects of nonlinearity, diffraction, and absorption on finite amplitude sound that passes through a focal region. Among the new results are power curves that show the exchange of energy between harmonics and beam patterns that show the location of nonlinearly generated sidelobes in the focal plane. [Work supported by ONR.]

2:15

J4. Finite amplitude acoustic waves generated by a baffled, multiharmonic transducer. James B. Edgerton, Jr. and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Arbitrary periodic excitation of a baffled piston transducer leads to harmonic and intermodulation distortion when finite amplitude effects are considered. An analysis of the radiation signal in such a situation is initiated by representing the linearized signal as a sum of King integrals associated with each harmonic in the excitation. A nonuniformly accurate expression for the velocity potential is derived using a singular perturbation procedure that identifies the dominant tendency of nonlinearity to generate second-order interactions. This form is obtained from an asymptotic integration, which leads to spectral decomposition of the signal into groups of quasi-one-dimensional waves. The renormalization version of the method of strained coordinates is used to obtain a uniformly accurate expression for the acoustic pressure at all locations preceding the formation of a shock. [Work supported by the ONR, Code 1125-UA.]

2:30

J5. Temperature dependence of the ultrasonic nonlinearity parameters of NaCl. Wenhwa Jiang, M. A. Breazeale (Department of Physics, University of Tennessee, Knoxville, TN 37996), and Ali Kashkooli (Department of Physics, Appalachian State University, Boone, NC 28608)

The ultrasonic harmonic generation technique using a capacitive detector has been used to measure the ultrasonic nonlinearity parameters of sodium chloride single crystals between 300 and 77 K. Since NaCl is a typical ionic crystal, its ordinary elastic constants have been of interest to many physicists. Likewise, its nonlinear properties, its third-order elastic constants, and their temperature dependence have been of interest to solid-state theory. However, to date the large thermal expansion coefficient and the fragility of NaCl have thwarted measurement of the temperature dependence of its nonlinearity parameters. Differential expansion between NaCl and a bonded quartz transducer often has resulted in sample cleavage. A special technique for bonding the quartz transducer to the NaCl sample has been developed. With the aid of this technique, ultrasonic nonlinearity parameters of three different oriented samples of NaCl have been measured for the first time in the temperature ranges of 300 to 77 K for the (100) sample, 300 to 100 K for the (110) sample, and 300 to 110 K for the (111) sample. The TOE constant C_{111} and two combinations of the other TOE constants and their temperature dependence have been obtained. The results are compared with theory describing interatomic forces in alkali halide crystals. [Research supported in part by the ONR and the UT-ORNL Science Alliance.]

2:45

J6. Dissipative structure of shock waves in fluids having large specific heat. M. S. Cramer (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

Recent studies indicate that the nonlinearity parameter $1 + B/2A$ may become negative in fluids whose specific heats are sufficiently large. The present study examines the dissipative structure of weak shocks in such fluids and contrasts the results obtained with those of the classical Taylor structure. Estimates for the thickness will also be presented. Conditions under which the thickness increases, rather than decreases, with increasing shock strength will be given.

3:00

J7. Ultrasonic absorption in the critical mixture of methanol and cyclohexane. Steven J. Fast, S. S. Yun, and F. B. Stumpf (Department of Physics and Astronomy, Ohio University, Athens, OH 45701)

The ultrasonic data for the critical mixture of methanol and cyclohexane are analyzed using the dynamic scaling theory. The experimental value of the slope of α/f^2 at the critical temperature versus $f^{-1.06}$ is compared to the theoretical value. The plot of α/f^2 at the critical temperature versus $f^{-1.06}$ yields a straight line as predicted by theory. The experimental values of α/α_c for methanol and cyclohexane are compared to the scaling function $F(\omega^*)$ and to the values of α/α_c of ^{3}He and Xe.

3:15

J8. Anomalies in the scattering-induced attenuation of backscattered ultrasonic waves. Peter B. Nagy and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

Scattering-induced ultrasonic attenuation offers a simple way to characterize quantitatively material inhomogeneities. This method has a wide range of applications from tissue characterization to ultrasonic NDE such as grain size measurement in polycrystalline materials and porosity assessment in cast metals. The scattering-induced attenuation of a through-transmitted well-collimated ultrasonic wave can be readily related to certain characteristics of the inhomogeneity via its total scattering cross section. In many cases, ultrasonic attenuation measurement is not feasible except from the backscattered signal. For want of better approximation, the scattering-induced attenuation is presumed to have the same relation to the inhomogeneity as if it were measured by the simpler transmission technique. Experimental results are presented to show that the backscattered signal is much less attenuated than is predicted by the plane-wave approximation, and that this effect is not simply due to multiple scattering. Furthermore, it is shown that the scattering-induced attenuation of the incoherent backscattered signal is mainly due to backward scattering, while strongly forward scattering inhomogeneities, such as surface roughness, will cause negligible attenuation. [Work sponsored by the Air Force Wright Aeronautical Laboratories/Materials Laboratory under Contract No. W-7405-ENG-82 with Iowa State University.]

3:30

J9. Effects of fiber motion on the acoustical behavior of an anisotropic, flexible fibrous material. Milo D. Dahl, Edward J. Rice, and Donald E. Groesbeck (National Aeronautics and Space Administration, Lewis Research Center, Cleveland, OH 44135)

The acoustic behavior of a flexible fibrous material was studied experimentally. The material consisted of cylindrically shaped fibers arranged in a batting with the fibers primarily aligned parallel to the face of the batting. This type of material was considered anisotropic with the propagation constant depending on whether the direction of sound propagation was parallel or normal to the fiber arrangement. Normal incidence sound absorption measurements were taken over the frequency range of 140–1500 Hz and with bulk densities ranging from 0.0046–0.067 g/cm^3. When the sound propagated in a direction normal to the fiber alignment, the measured sound absorption showed the occurrence of a strong resonance that increased absorption above that attributed to viscous and thermal effects. A model for the material indicated that this resonance was due to fiber motion. When the sound propagated in a direction parallel to the fiber alignment, indications of the additional sound absorption due to fiber motion were not present in the data.

3:45

J10. Effect of ambient pressure on the pressure wave from the rapidly expanding bubble. Ho-Young Kwak[a] (Sibley School of Mechanical and Aerospace Engineering, Cornell University, Ithaca, NY 14853)

Calculation on the intensity of evaporation and the rapidly expanding bubble formed from the droplet at the superheat limit is calculated. An analysis is presented for the amplitude of the pressure wave generated by the rapidly expanding bubble. For estimating the amplitude, linear acoustics theory for the evaporating droplet and the Rayleigh equation for the expanding bubble are employed. The result shows that the amplitude and the frequency of the pressure wave depend crucially on the difference in pressure between vapor within the bubble and the surrounding liquid. This result is in qualitative agreement with previous experiment. [a] On leave of absence from Chung-Ang University, Seoul, Korea.

4:00

J11. Forced radial oscillations of single cavitation bubbles: Period-doubling, chaos, and hysteresis. R. G. Holt and L. A. Crum (Physical Acoustics Research Laboratory, Department of Physics, University of Mississippi, Oxford, MS 38677)

Utilizing an optical scattering technique previously reported [J. Acoust. Soc. Am. Suppl. 1 80, S24 (1986)], the radial response of a single bubble to an acoustic field has been observed and analyzed. A scattered light intensity versus radius transfer function has been obtained for the system, enabling a quantitative description of the radius versus time. Some universal features of driven nonlinear systems have been observed, among them period-doubling, chaotic solutions (with corresponding fractal attracting sets in the phase plane), and bistability due to hysteresis in the resonance structure. [Work supported by ONR.]

Session K. Psychological and Physiological Acoustics III: Binaural Processing

Gary L. Gibian, Chairman
Department of Physics, American University, 4400 Massachusetts Avenue N.W., Washington, DC 20016

Contributed Papers

1:30

K1. Predictions for binaural masked detection with frozen-noise maskers. Gary C. Kline and H. Steven Colburn (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139 and Biomedical Engineering Department, 110 Cummington Street, Boston University, Boston, MA 02215)

A model of binaural hearing based on combinations of interaural time and intensity differences [Gabriel *et al.*, J. Acoust. Soc. Am. Suppl. 1 **74**, S85 (1983)] was used to predict results obtained previously [R. H. Gilkey, D. E. Robinson, and T. E. Hanna, J. Acoust. Soc. Am. **78**, 1207–1219 (1985)] in a binaural masked detection experiment with repeatable samples of masking noise. The narrow-band model generates its decision variables through a linear combination of interaural time and intensity differences followed by a sluggishness filter (averager). A computer simulation of the model was used to predict the hit and false-alarm rates for each of ten independent, frozen-noise masker samples, and for target phase angles of 0 deg and 90 deg. Model predictions disagree with the experimental results in two respects. First, the false-alarm rates predicted by the model show almost no variation across masker samples, while the experimental results show changes from 5% to 80%. This disagreement illustrates the inadequacy of the additive internal noise assumption of the model. Second, the model predicts changes in the hit rate as large as 80% when the tone phase changes from 0 to 90 deg for a given masker sample, while the experimental results show changes less than 20% for a given masker sample (although hit rates vary over a range of 75% across masker samples). This significant correlation in hit rates for these two phase angles is not easily explained within a model based on interaural differences since the interaural differences in the stimulus are uncorrelated for these phase angles. [Work supported by NIH.]

1:45

K2. Extending the position-variable model: Dependence of lateralization on frequency and bandwidth. Glenn D. Shear and Richard M. Stern (Department of Electrical and Computer Engineering, Carnegie–Mellon University, Pittsburgh, PA 15213)

The position-variable model [R. M. Stern, Jr. and H. S. Colburn, J. Acoust. Soc. Am. **64**, 127–140 (1978)] is extended to describe the subjective lateral position of tones as a function of frequency and ITD, and the lateralization of bandpass noise as a function of bandwidth, ITD, and interaural phase shift. The two major modifications made to the model are (1) the use of a frequency-dependent form of the function that expresses the relative number of binaural coincidence detectors as a function of their characteristic delay and (2) additional processing of the display of interaural cross correlation to emphasize those regions of the function which exhibit peaks at the same internal delay across a range of frequencies. For most naturally occurring broadband stimuli, these regions indicate the set of primary modes of the corresponding cross-correlation functions, thereby enabling the central processor to determine without ambiguity the value of ITD present in the stimulus. The extended model describes lateralization data for a much wider range of stimulus frequencies than the original model, which could only describe responses elicited by 500-Hz tones. [Work supported by NIH.]

2:00

K3. The dependence of binaural detection and interaural discrimination on interaural time and intensity in normal and impaired listeners. J. Koehnke and H. S. Colburn (Research Laboratory of Electronics, MIT, Cambridge, MA 02139 and Boston University, 100 Cummington Street, Boston, MA 02215)

Measurements of binaural performance by normal and hearing-impaired listeners are reported for large reference values of interaural time delay (ITD) and interaural intensity difference (IID). NoSπ detection thresholds and just-noticeable differences (jnds) in ITD and IID were measured with 1/3-oct noise bursts at 500 and 4000 Hz, using reference ITDs and IIDs up to 600 μs and 34 dB. As expected, jnds and thresholds for normal and impaired subjects increase as the reference IID increases. In contrast, increasing the reference ITD shows relatively small effects for most subjects. The present data are consistent with those described previously [C. Passaro, J. Koehnke, and H. S. Colburn, J. Acoust. Soc. Am. Suppl. 1 79, S21–S22 (1986)] but are more pronounced with larger reference ITDs and IIDs. For example, performance is generally poorer for the impaired listeners than the normal listeners. Also, for both groups, measurements of IID jnds consistently result in smaller jnds for the canceling combinations of reference ITD and IID than reinforcing combinations. [Work supported by NIH.]

2:15

K4. Nonlinear spectral interferometry (NLSI): A new approach to modeling binaural hearing and aural communication channeling. Nathan Cohen (Department of Natural Sciences, Bentley College, Waltham, MA 02254) and Dean Cummins (Harvard Biological Laboratories, Harvard University, Cambridge, MA 02138)

A biointerferometric model for spatial hearing is described in which binaural information is processed as an interferometric response via a nonlinear spectral transform of the cross-correlation function. Relevant observables include phase delay, phase delay rate, visibility amplitude, and visibility phase. Our model accounts for a broad range of hearing phenomena including: (1) localization below diffraction limited frequencies by overtone generation at the cochlea; (2) localization acuity by superresolution of the synthesized beam; (3) beam synthesis by visibility plane sampling through overtone content in speech and cochlear response; (4) the binaural advantage in intelligibility through interferometric rejection and neuronally driven beam steering; and (5) transformation from phase to amplitude localization cues through loss of visibility phase above the phase-locked frequency range. This model obviates the need for multiple cue and/or component models and has distinct advantages over other cross-correlation models. Additional psychoacoustic tests to corroborate this NLSI model are described.

2:30

K5. The effect of head-induced interaural time and level differences on speech intelligibility in noise. A. W. Bronkhorst and R. Plomp (Department of Otolaryngology, Free University Hospital, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands)

The effect of interaural time delay (ITD) and acoustic headshadow on binaural speech intelligibility in noise was studied. A free-field condition was simulated by presenting recordings, made with a KEMAR manikin in an anechoic room, through headphones. Recordings were made of speech, reproduced in front of the manikin, and of noise, emanating from seven angles in the azimuthal plane, ranging from 0° (frontal) to 180° in steps of 30°. From this noise, two signals were derived, one containing only ITD, the other containing only headshadow. Speech-reception thresholds for sentences in noise for a group of normal-hearing subjects showed that for noise azimuths between 30° and 150°, the gain due to ITD lies between 3.9 and 5.1 dB, while the gain due to headshadow ranges from 3.5 to 7.8 dB. A second experiment with similar stimuli, presented monaurally or with a 20-dB interaural level difference, indicated that for noise with only headshadow, the gain relies on the ear presented with the most favorable signal-to-noise ratio, but decreases when the noise presented to the other ear becomes relatively loud.

2:45

K6. Motion aftereffects with horizontally moving sound sources. D. Wesley Grantham and Lynn E. Luethke (Bill Wilkerson Hearing and Speech Center, 1114 19th Avenue South, Nashville, TN 37212)

Continuous sounds were presented in an anechoic chamber through two horizontally rotating loudspeakers that traversed a full 360° around the observer at ear level (distance: 1.5 m). At 10-s intervals this "adaptation stimulus" was interrupted and a 750-ms "probe stimulus" was presented from a pair of stationary loudspeakers (separated by 7.5°) 1.6 m in front of the observer. The probe could itself be stationary or could "move" (employing a stereophonic balancing algorithm) in either direction. During a run the adaptation stimulus was held at a constant velocity (− 200° to + 200°/s), while probes with velocities varying from − 10° to + 10°/s were presented in a random order. Observers judged the direction of motion ("left" or "right") of each probe tone. When the frequency content of the adaptation stimulus was the same as that of the probe stimulus (either 500-Hz low-pass or 6300-Hz high-pass noise), stationary probes were consistently judged to move in the direction opposite to that of the adaptation stimulus. This effect increased with the velocity of the adaptor. Possible sensory and nonsensory mechanisms underlying this aftereffect will be discussed. [Work supported by NIH.]

3:00

K7. A comparison of the effects of time and intensity during the auditory brain-stem response to lateralized clicks. Mark Stephenson (Armstrong Aerospace Medical Research Laboratory, Biodynamics and Bioengineering Division, Biological Acoustics Branch, Wright–Patterson AFB, OH 45433) and William Melnick (Department of Otolaryngology, The Ohio State University, Columbus OH 43210)

A group of 12 otologically normal subjects were presented with binaural clicks at 70 dB nHL. During one condition, subjects delayed the onset of the click to the left ear until a single image was percieved either midway between the center of the head and the right ear (referred to as the "midway" location), or until the image was perceived just at the right ear. In a second condition, subjects attenuated the click to the left ear until the acoustic image was lateralized either midway towards the right ear, or just at the right ear. These same stimuli conditions were then used to evoke auditory brain-stem responses (ABRs). When clicks were lateralized midway, ABR latencies from delayed versus attenuated clicks were not significantly different. However, when clicks were lateralized just at the right ear, there were statistically significant differences between latencies from delayed versus attenuated clicks. There was also a consistent ordinal relationship noted: In nearly every case, delayed clicks yielded longer latencies than attenuated clicks. The clear perceptual equivalence observed psychoacoustically was not fully reflected in the corresponding auditory brain-stem responses. [Work supported by AFOSR.]

3:15

K8. The duplex nature of the brain-stem binaural interaction component: Frequency, rate, and intensity effects. T. K. Parthasarathy and G. Moushegian (Communication Disorders, Callier Center, University of Texas at Dallas, Dallas, TX 75235)

As in a previous study [J. Acoust. Soc. Am. Suppl. 1 79, S6 (1986)], stimuli were varied to evaluate the binaural interaction component (BIC) of the auditory brain-stem response (ABR) in normal hearing adults. In addition to a clicklike sound, tone bursts (0.5, 1.0, and 2.0 kHz), having two cycles of rise–fall time and one of duration, were presented at two intensities (85 and 100 dB SPL). Binaural wave III and V amplitudes were smaller with shorter latencies than summed monaural amplitudes and latencies at both intensity levels of the click and 2.0-kHz tone burst. Increasing stimulus rate produced longer latencies and smaller peak amplitudes of the binaural interaction components and concomitantly longer latencies and smaller wave III and V amplitudes of ABR. Wave V latency shift was greater at 85 than at 100 dB SPL, whereas the BIC exhibited greater latency shifts at 100 than at 85 dB SPL. The *derived* frequency following response (FFR) to 0.5 and 1.0 kHz [J. Acoust. Soc. Am. Suppl. 1 79, S6 (1986)] had shorter latencies and larger amplitudes at 100 than at 85 dB SPL. All of the findings suggest that the BIC, in conformity with the findings from medullary neuronal physiology, is differentially sensitive to rate, frequency, and intensity.

3:30

K9. Sound localization in the budgerigar and the interaural pathways. Thomas Park, Kazuo Okanoya, and Robert Dooling (Department of Psychology, University of Maryland, College Park, MD 20742)

Sound localization presents a problem for animals with small heads (closely spaced ears) and poor high-frequency hearing. In recent tests of spatial resolving power, several small songbirds (i.e., great tit, canary, and zebra finch) show rather large minimum audible angles (MAA) on the order of 15–20 deg. These results are expected if conventional binaural time and intensity differences are used. Budgerigars (17-mm interaural distance), on the other hand, demonstrate an MAA of about 5 deg, which approaches the excellent ability of the barn owl with a much larger head (50-mm interaural distance). Recent work has suggested that interaural pathways connecting the two ears of birds could be involved in sound localization. Using latex injection medium, we have demonstrated interaural pathways in the budgerigar, canary, and zebra finch. Taken together, the results of these behavioral and anatomical studies indicate that the budgerigar may use the interaural pathway in sound localization.

3:45

K10. Performance of adult subjects on a dichotic speech test under both directed and free recall listening conditions. Jane A. Baran (Department of Communication Disorders, University of Massachusetts, Amherst, MA 01003 and Dartmouth–Hitchcock Medical Center, Hanover, NH 03756) and Frank E. Musiek (Dartmouth–Hitchcock Medical Center, Hanover, NH 03756)

Twenty-five young adult subjects with negative otologic histories were administered a dichotic rhyme test under three different listening conditions: (1) free recall, (2) directed listening to the right ear, and (3) directed listening to the left ear. The dichotic rhyme test used is composed of 30 well-aligned synthetic CVC words that were presented at 50 dB SL (*re*: speech reception thresholds). The nature of the test is such that under normal conditions (i.e., free recall), listeners tend to repeat either the word presented to the left ear or to the right ear. Normal performance is approximately 50% correct identification in each ear, with a slight right ear advantage evident. In an earlier investigation using a dichotic CV test, Keith *et al.* [Ear Hear. 6, 270–273 (1985)] demonstrated a clear left ear advantage on a directed left ear task and an obvious right ear advantage on a directed right ear task. In the present investigation, no significant differences in the test scores were observed when the right and left ear scores were compared with the same ear scores across the three test conditions.

In all three test conditions, a slight right ear advantage was noted. The implications of these findings as they relate to our understanding of how dichotic stimuli are processed will be discussed.

4:00

K11. Monaural perception of the rapidly alternating speech perception test (RASP). Richard W. Harris, Ruth Kaspar, and Robert H. Brey (131 TLRB, Communication Sciences and Disorders Area, Brigham Young University, Provo, UT 84602)

One test that claims to measure binaural fusion, and is routinely utilized as a measure of central auditory function, is the rapidly alternating speech perception (RASP) test. There has been some controversy over the clinical validity of this test. The purpose of this study was to investigate information contained in a single channel of the RASP. Twenty-four normal hearing subjects listened to one channel of the RASP: twelve to channel 1, twelve to channel 2. Sentence scores for a single channel of the RASP ranged from 0% to 70%, with mean sentence scores for subjects listening to channels 1 and 2 of 37.72% and 20.8%, respectively. An ideal binaural fusion test would contain no, or very little, information when listening to only one channel. It is possible that a single channel of the RASP contains too much information. Many of the sentences were repeated correctly, in their entirety, by at least a few of the subjects listening to either channel in isolation, a task previously assumed to be possible only in the binaural condition.

TUESDAY AFTERNOON, 12 MAY 1987 — CANYON HALL, 2:00 TO 5:00 P.M.

Session L. Musical Acoustics I: Demonstration of Computer-Based Piano Performance Reproduction System

Uwe J. Hansen, Chairman
Indiana State University, Terre Haute, Indiana 47809

Chairman's Introduction—2:00

Invited Paper

2:05

L1. Computer-based piano performance reproduction system. Hal Vincent (Kimball World, P. O. Box 460, Jasper, IN 47546)

This session will consist of a demonstration of the Bosendorfer 225SE computer-based piano performance reproduction system. The system incorporates an optical sensing mechanism to monitor and record individual key positions for subsequent exact reproduction of performance dynamics. The demonstration will take approximately 1 h, followed by a 1/2-h question and answer period. It will be repeated again from 3:30–5:00 p.m., and again on Wednesday from 10:00–11:30 a.m., and on Friday from 9:00–10:30 a.m. The instrument will also be available, by arrangement, for small group hands-on evaluations.

Session M. Architectural Acoustics II: Computers in Architectural Acoustics

David Braslau, Chairman
David Braslau Associates Incorporated, 1313 5th Street S.E., Suite 322, Minneapolis, Minnesota 55414

Chaiman's Introduction—2:30

Invited Papers

2:35

M1. Computers in architectural acoustics. Manfred R. Schroeder (Drittes Physikalisches Institut, Universität Göttingen, Federal Republic of Germany and AT&T Bell Laboratories, Murray Hill, NJ 07974)

Already in the 1950s, difficult problems in acoustics were being successfully attacked by computer simulation. One of the first digital simulations of a signal processor was that of an artificial reverberator to produce natural-sounding reverberation. Other early applications of computers used Monte-Carlo simulations to study random wave interference in enclosures. Later, computers were used extensively in the measurements in Philharmonic Hall (now Avery Fisher Hall) in New York City. This work also led to the development of new methods for measuring reverberation time. In the 1970s, with the support of the German Science Foundation, a large number of concert halls were investigated with the help of sound field reproduction and multidimensional scaling methods using digital computers. The most important new parameter emerging from these studies was "interaural dissimilarity." The diffusely reflecting surfaces called for by these results are based on number-theoretic principles [M. R. Schroeder, *Number Theory in Science and Communication* (Springer, New York, 1986) 2nd enlarged ed.]. Computers have also been used to crack complicated integral equations to yield accurate relationships between reverberation time and sound absorption that are sensitive to absorber location and sound diffusion.

3:05

M2. Prediction of point-to-point acoustic transfer function within a three-dimensional enclosure using ray path methods. James L. Wayman,[a] Robert V. Esperti (Delco Systems Operations, MS 101, Goleta, CA 93117), and James P. Vanyo (Department of Mechanical and Environmental Engineering, University of California, Santa Barbara, CA 93106)

This report details a study conducted for the General Motors Corporation to construct a model for predicting point-to-point acoustic transfer functions in automobile passenger compartments for the purpose of predicting entertainment system performance. The model is based on ray path methods and includes effects of curved enclosure surfaces and complex boundary impedances. Comparisons between model results and experimentally obtained transfer functions are promising. [a] Now with Ford Aerospace and Communications Corporation, Systems Technology Development Department, X-21, 3939 Fabian Way, Palo Alto, CA 94303.

3:30

M3. Determination of the direction, time, and intensity of arrival of acoustic signals. Farrel M. Becker (Audio Artistry, P. O. Box 56, Kensington, MD 20895)

There is currently much interest in lateral reflections in concert halls. Attempts to quantify lateral reflections have, up to now, been limited. Lateral energy ratios have been measured using a bidirectional microphone. This technique lacks the resolution necessary to investigate direction other than side to side versus front to back. A new technique, known as the polar energy time curve, will be presented. Utilizing special software for equipment already in common use, this technique extracts accurate directional information from four energy time curve (or impulse response) measurements made with an ordinary cardioid microphone. The direction of arrival versus time or intensity is displayed in polar form. The time, intensity, and exact *angle* of arrival may be determined in any plane of interest. Three-dimensional displays of energy arrival are also possible. Conventional lateral energy fractions, as well as other quantities, may be computed.

3:55

M4. Evaluation of room speech transmission index and modulation transfer function by the use of time delay spectrometry. D. B. Keele, Jr. and Donald Eger (Techron, Division of Crown International, Inc., 1718 West Mishawaka Road, Elkhart, IN 46517)

The literature shows that the modulation transfer function (MTF) and speech transmission index (STI) can be computed from the squared impulse response of a linear passive system. This paper describes an extension of this method to measurements of systems using time delay spectrometry (TDS). The new method makes use of both the real and imaginary parts of the complex analytic impulse response of the system (the energy-time response). This allows more accurate determinations of STI and MTF because the calculations are based on a measurement that more closely follows the true energy decay in the room. The method requires fewer samples of the room's sound field and less spatial averaging to yield a given accuracy.

4:15

M5. Criteria and analysis for fan/floor vibration isolation in elevated mechanical rooms. Angelo J. Campanella (Campanella Associates, 3201 Ridgewood Drive, Columbus, OH 43220)

In selecting isolators for fans and other mechanical equipment mounted on elevated mechanical room floors, it is necessary to know the degree with which their vibrations will affect the occupants of the floor on which it is mounted, as well as the floor below. A computational procedure was developed based on a two degree of freedom model representing the resulting floor motion for known masses, resonances, and damping of the floor and fan mount. A computational criterion was developed for vibration input and human vibration tolerance based on values commonly found in the literature. A FORTRAN program featuring interrogatory input prompts and a screen graphics output of the floor vibrational velocity over the frequency band of concern executes in about 30 s. With this tool, one can immediately judge the effectiveness of various fan masses and spring constants. Two case studies are discussed.

TUESDAY PM

Contributed Papers

4:35

M6. A presentation of acoustical impulse responses with an analog-to-digital converter. Edward G. Clautice (Department of Architecture, 231 ARCH, University of Florida, Gainesville, FL 32611)

A research program in architectural acoustics at the University of Florida has been using analog electrical equipment to process signals. An analog-to-digital converter was acquired to allow faster and better signal analysis. Several computer programs were written that control the analog- to-digital converter and subsequent processing of the digital data. The ability of the analog-to-digital converter to analyze some properties of known signals was evaluated. The analog-to-digital converter was then used to evaluate signals from magnetic tapes which had the recorded acoustical impulse response from several positions in a 1500 seat auditorium. The processing programs were used to produce echograms. Those digital echograms were able to duplicate the echograms produced by the laboratory's existing instrumentation in terms of the relative location of events in time and the decay process through time. The process and results of this research will be surveyed.

4:50

M7. The use of an IBM personal computer (PC) for performing room acoustics measurements. Richard H. Talaske (R. L. Kirkegaard & Associates, Inc., P. O. Box 186, Downers Grove, IL 60515)

The general acceptance of the PC microcomputer and the subsequent availability of additional hardware and software have offered researchers and practitioners some formidable tools for acoustical investigations. Using an IBM portable PC, ILS signal-processing software, and additional signal gathering and signal conditioning equipment, an inexpensive and portable room acoustics and data collection system was developed for the purpose of investigating the acoustics of theaters, concert halls, and other assembly spaces. This system will be described, and the results of data will be presented and discussed.

Session N. Education in Acoustics: Modeling in Acoustics Education

Gary W. Siebein, Chairman
Department of Architecture, University of Florida, Gainesville, Florida 32611

Chaiman's Introduction—8:00

Invited Paper

8:05

N1. Acoustical scale modeling in the classroom at MIT. Richard G. Cann (Grozier Technical Systems, Inc., 157 Salisbury Road, Brookline, MA 02146)

This is an actual demonstration of the acoustical scale modeling methods that have been used by the author for several years to help teach some basic acoustics to students in two classes at the Department of Architecture. One class in the curriculum of Building Technology, instructed by Carl Rosenberg of BBN, uses scale modeling to show how the reverberant sound within an interior space forms from numerous wall reflections. Echograms from various model room configurations illustrate simple defects that may occur in the quality of the interior sound. Also, students see demonstrated clearly the difference between the reduction of noise by surface absorption and by transmission loss. Another course attended by equal numbers of undergraduate and graduate students is instructed by Tim Johnson and John Furlong. It explores various aspects of the microclimate. One part, acoustics, is presented in the form of one lecture and a brief acoustical scale modeling demonstration immediately followed by a laboratory activity in which students investigate the control of sound from a highway into a nearby community. Students use the scale modeling instrumentation to help select the appropriate noise barrier that will give an acceptable community noise level, or/and noise reduction, at the lowest cost. There is no one solution to the problem, but students must show their understanding of acoustics by justifying their selection quantitatively.

8:50–9:00

Break

Contributed Papers

9:00

N2. A method to evaluate the acoustical effects of design decisions in the architectural studio class. Gary W. Siebein (Department of Architecture, 231 ARC, University of Florida, Gainesville, FL 32611)

The results of a research project in acoustical modeling conducted by the author were used to facilitate the investigation of acoustical issues in the conceptual stage of the design process of students in a fourth-year architectural studio. The students worked on a semester-long project to design a multipurpose performing arts center. The intent of the studio was to look at the potential of acoustics as a formgiver in architecture. The acoustical modeling technology was used as a mechanism to evaluate the proposed solutions. The technique allowed students to receive immediate feedback on the acoustical quality of a proposed scheme. The results of tests in models of the students' designs were compared to the results of similar tests made in many different auditoria gathered as a part of the research project. Students were able to test many alternative design schemes and understand the resulting changes in acoustical quality (as evaluated by new objective design criteria such as early to late temporal energy ratios and reflectograms) in small-size study models. Once a basic solution was reached, more detailed architectural and acoustical refinement of the scheme followed. The high energy level and enthusiasm of the students and the most satisfactory design projects at the end of the semester attested to the success of this new design tool. [Work supported by NSF.]

9:15

N3. A microcomputer-based demonstration system for musical acoustics education. Robert Maher and James Beauchamp (School of Music, University of Illinois at Urbana—Champaign, 2136 Music Building, 1114 West Nevada, Urbana, IL 61801)

While an ideal musical acoustics education laboratory would include an extensive collection of electronic measurement, recording, and display equipment together with appropriate space, staff, and maintenance staff to support student use, this situation is seldom possible because of limited funds. Our solution has been to configure and program a microcomputer (IBM PC AT) to serve as a mobile multifunction demonstration station. The station is capable of high-resolution color display, stereo audio/digital input/output, hardware FFT analysis, and MIDI input/output. Thus far, the following demonstrations have been completed: (1) animated display simulation of wave propagation on strings and within pipes; (2) waveform/spectrum display and sound synthesis from given Fourier amplitudes and phases; (3) single frame spectrum analysis of audio signals; (4) display simulation of modes on square and circular vibrating membranes; and (5) sound synthesis comparison of just, Pythagorean, and equal-tempered scales. One advantage of this microcomputer-based system is that new demonstrations can be incorporated easily by writing new programs for the existing hardware. [Work supported by an IBM grant to UIUC.]

Session O. Psychological and Physiological Acoustics IV: Complex Signals

Marjorie R. Leek, Chairman
University of Minnesota, Department of Communication Disorders, 115 Shevlin Hall, 164 Pillsbury Drive S.E., Minneapolis, Minnesota 55455

Contributed Papers

8:30

O1. Detection of time-varying sequential constraints in finite-state auditory pulse trains. Irwin Pollack (Mental Health Research Institute, University of Michigan, Ann Arbor, MI 48109)

Listeners discriminated between auditory pulse trains generated by (a) unconstrained finite-state random sequences and (b) statistically constrained finite-state sequences. The nature of the statistical constraint was to make one half of the possible set of successive *n*-tuples more likely than the remaining half. The statistical constraint was changed periodically and aperiodically about an average zero constraint level. Discrimination performance deteriorates as the statistical constraint is changed more frequently, as might be anticipated with a fixed informational integration period or "auditory moment."

8:45

O2. The effects of decorrelation on the discriminability of noise samples. Susan M. Fallon and Donald E. Robinson (Department of Psychology, Indiana University, Bloomington, IN 47405)

In a previous study, it was demonstrated, using a same–different paradigm, that the discriminability of samples of reproducible noise is a function of the correlation between members of a pair of samples to be discriminated [S. M. Fallon and D. E. Robinson, J. Acoust. Soc. Am. Suppl. 1 **78**, S46 (1985)]. On "different" trials, the second sample of a pair was generated by repeating a temporal segment of the sample presented in the first interval, and combining it with a new segment of noise. The data from that experiment indicate that discriminability at "threshold" is independent of the total duration of the sample, but is highly dependent on the interpair correlation (proportion of common elements). Discriminability was also affected by the temporal position of the new segment of noise. In the work reported here, the correlation (proportion of common variance) between a pair of identical samples was varied by adding a new, independent sample of noise to the sample presented during the first observation interval. The duration and position of the added sample were also varied. The similarities and differences between the two methods used to decorrelate the pairs of samples will be discussed. [Work supported by AFOSR and NMRDC.]

9:00

O3. Perception of multidimensional complex sounds. Gary R. Kidd and Charles S. Watson (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Listeners' abilities to perceive information independently encoded in different dimensions of complex sounds were examined in experiments that required simultaneous attention to three dimensions. Stimuli consisted of sequences of one, three, five, or seven brief pulses that were generated by adding five 100-ms sinusoidal components. Each pulse had one of two values on each of the following complex dimensions: (1) harmonicity (harmonic versus inharmonic relations among the components); (2) spectral shape (linearly decreasing amplitude versus a two-peaked amplitude profile); and (3) amplitude envelope (slow versus rapid rise and decay times). Stimuli were selected such that the two values on each dimension were highly discriminable. In a classification task involving combinations of dimensional values, listeners showed an impressive ability to integrate information over pulses within sequences, with performance at about 5% to 10% below ideal performance for all sequence lengths. However, performance was not a simple function of the discriminability of the individual dimensions, and there were major differences among listeners' tendencies to differentially attend to each of the dimensions. [Work supported by USN/NMRDC and by AFOSR.]

9:15

O4. Multiple observations and internal noise. B. G. Berg and D. E. Robinson (Department of Psychology, Indiana University, Bloomington, IN 47401)

A multiple observation task was used to evaluate models of "internal noise." On each trial, n tones ($n = 1,2,3,4,6,8,10$, or 12 within a block) were independently sampled from one of two probability density functions on frequency; each was normal in form with a standard deviation of 100 Hz. The two distributions had means of 1000 and 1100 Hz, respectively. The sample was presented through headphones as n, 50-ms tone bursts separated by 50 ms. The subjects' task was to decide from which of the two distributions the tones were sampled. A model partitioning internal variance into peripheral variance, added to each observation prior to formulating a decision statistic, and central variance, added to the decision statistic, provided a good description of the data of all four subjects. A subsequent study counterbalanced two levels of the difference between distribution means ($\Delta\mu = 100$ or 150 Hz) with two levels of the standard deviation ($\sigma = 100$ or 150 Hz). Increasing $\Delta\mu$ did not affect estimates of internal variance. However, increasing σ resulted in increased estimates of internal variance for both levels of $\Delta\mu$. One interpretation of this result is that external and internal variance are not independent. [Work supported by AFOSR and NMRDC.]

9:30

O5. Frequency response and perceived sound quality. Alf Gabrielsson, Björn Hagerman, and Ove Till (Department of Technical Audiology, Karolinska Institute, KTH, S-10044 Stockholm, Sweden)

The perceived sound quality of sound-reproducing systems can be described in terms of perceptual dimensions as clarity, fullness, brightness, spaciousness, and others [A. Gabrielsson and H. Sjögren, J. Acoust. Soc. Am. **65**, 1019–1033 (1979)], and fidelity may be considered as a weighted combination of the perceptual variables. The frequency response probably affects all these variables. An experiment, in which the frequency response was manipulated, showed that, among other things, brightness and sharpness increased with increased frequency response at higher frequencies, fullness and nearness decreased with decreased response at lower frequencies, and clarity increased with a certain rise above 1 kHz in comparison with a flat response. The results were more pronounced for normal hearing than for hearing impaired persons, but both categories preferred a frequency response increasing above 1 kHz to a flat response. Similar results were obtained from an analysis of the frequency response of high-fidelity loudspeakers used in an extensive listening test [A. Gabrielsson and B. Lindström, J. Aud. Eng. Soc. **33**, 33–53 (1985)].

This was especially evident when the frequency response was measured directly in the listening room.

9:45

O6. Temporal resolution in preschool children. F. L. Wightman, T. R. Dolan, P. Allen, and D. Jamieson (Waisman Center on Mental Retardation and Human Development, University of Wisconsin, Madison, WI 53705)

An adaptive three-alternative forced-choice paradigm, in the form of a video game, was used to test auditory temporal processing skills in young children. Thirty children between the ages of three and seven were asked to detect a temporal gap in a burst of octave band noise, centered at 400 or 2000 Hz. The minimum detectable gap (gap threshold) was estimated by taking the median of the threshold estimates from at least three 20-trial adaptive runs. While the variability in the median threshold estimates was substantial, the data suggest that low-frequency gap thresholds were higher for the children than for adults, with the younger children producing higher thresholds. High-frequency gap thresholds were comparable to adult values for all children. All the individual adaptive runs were "adult-like," suggesting that the subjects were concentrating on the task. However, run-to-run variability, as well as subject-to-subject variability, was high. It seemed that the children had "good days" and "bad days," and that the younger children were more influenced by these variations than their older colleagues. [Work supported by NICHD Grant 5 P30 HD03352.]

10:00

O7. Detection of envelope phase disparity. Gregory H. Wakefield (Department of Electrical Engineering and Computer Science, University of Michigan, Ann Arbor, MI 48109)

The sensitivity of the auditory system to temporal variations across frequency in the time-varying spectrum of a signal by measuring the detectability of a phase disparity between the envelopes of two sinusoidally amplitude-modulated carriers is discussed. Signals were 500 ms in duration and were gated by a 30-ms linear window. A 2IFC procedure was used to obtain psychometric functions for relative phase between the envelopes of a 0.5- and 2-kHz carrier. The results indicate that the psychometric function is independent of modulation frequency from 8 to 64 Hz. A phase shift of approximately 45 deg is necessary for a d' of 1. Changes in the upper carrier frequency (e.g., 1 or 4 kHz) do not affect sensitivity. Furthermore, little change in performance is observed when additional SAM carriers are present. [Work supported by the EECS Department, Kresge Hearing Research Institute, and AFOSR.]

10:15

O8. Temporal integration of amplitude modulation. Stanley Sheft (Program in Audiology and Hearing Impairment, Northwestern University, Evanston, IL 60201)

The threshold for detection of sinusoidal amplitude modulation of a gated broadband noise carrier was measured for modulating signal durations varying from 12.5 to 400 ms. Stimuli were shaped by a 25.6-ms rise–fall time with modulation onset subsequent to shaping. Carrier level was first adjusted to compenste for change in rms due to modulation, then randomized ± 3 dB about a mean spectrum level of 25 dB SPL to ensure detection not based on change in overall level. As in the results of Viemeister [J. Acoust. Soc. Am. **66**, 1364–1380 (1979)], modulation transfer functions for gated noise carriers exhibited a bandpass characteristic. Addition of a 500-ms fringe proceeding modulation indicated that the high-pass segment was not due to fast-acting adaptation. For modulation frequencies above the high-pass segment, the amplitude of the modulating signal at threshold decreased roughly 7 to 8.5 dB per decade increase in modulation duration. Results suggest that the output of an auditory modulation detector is summed by a second integrator in a similar manner as seen in a broadband signal detection task.

10:30

O9. Evidence for sensitivity to rates of change of frequency and intensity. Gary J. Dooley and Brian C. J. Moore (Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England)

Thresholds for the detection of differences in duration were measured in a two-alternative forced-choice task for four types of signals, all centered at 2 kHz with a mean level of 65 dB SPL: (1) sinusoids fixed in frequency and level; (2) sinusoids of fixed frequency whose level was swept up or down by 5 or 10 dB; (3) sinusoids of fixed level whose frequency was swept up or down by 100 Hz; and (4) sinusoids whose level was swept up or down by 10 dB and whose frequency was swept up or down by 100 Hz. The duration of the standard was either fixed at 500 ms, or was varied randomly from trial to trial by up to $\pm 7\%$ about 500 ms. The pattern of results was similar for all four subjects tested. Thresholds for the signals which were swept in level and/or frequency were lower than those for the fixed signals, typically by 10 to 20 ms. This indicates that subjects were sensitive to the rate of change of frequency and/or intensity, and could use this as a cue for duration discrimination. [Work supported by the MRC, United Kingdom.]

10:45

O10. Perception of bird calls by budgerigars and humans: A multidimensional scaling analysis. Susan D. Brown and Robert J. Dooling (Department of Psychology, University of Maryland, College Park, MD 20742)

A combination of behavioral testing and multidimensional scaling analysis was used to test budgerigar perception of a natural vocalization: the "contact" call. For comparison, humans were tested on the same set of sounds. Individual differences multidimensional scaling (SINDSCAL) revealed that budgerigars and humans perceive budgerigar contact calls differently. Acoustic properties of the calls that covary along the dimensions determined by multidimensional scaling were identified. For humans, the absolute pitch of the call was most salient, while, for budgerigars, the spectral shape was most salient. These results are discussed in terms of the basic auditory capabilities of buderigars and humans and in terms of the coding of information in budgerigar vocal signals.

11:00

O11. When theory and practice collide: The use and abuse of statistics in bioacoustics. Leslie A. Wheeler (Laboratoire d'Acoustique Animale, E.P.H.E., INRA-CNRZ, F78350, Jouy-en-Josas, France and Massachusetts Institute of Technology, Room 36-511, Cambridge, MA 02139) and René-Guy Busnel (Laboratoire d'Acoustique Animale, E.P.H.E., INRA-CNRZ, F78350 Jouy-en-Josas, France)

All of us have been guilty of favoring statistical procedures that support our hypotheses. This is fine if our hypotheses are correct. But what if they are false? No matter. By use of appropriate statistical procedures, applied in just the right way, even the saggiest of hypotheses can be propped up. This paper explains how. As an example, the usefulness of 14 useless acoustic features in the differentiation and recognition of lovebird calls will be shown statistically in two different ways.

11:15

O12. English and prevoiced VOT (voice-onset-time) discrimination by monkeys and humans. Joan M. Sinnott and Frank S. Adams (Department of Psychology, Indiana University, Bloomington, IN 47405)

Discrimination of synthetic English and prevoiced VOT (voice-onset-time) continua was assessed in Old World monkeys and humans using a repeating standard AX procedure and positive-reinforcement operant conditioning techniques. The DLs were measured at VOTs of + 70 ms (English /pa/), 0 ms (English /ba/), and − 70 ms (prevoiced /mba/). Monkey DLs for /ba/ (standard)–/pa/ (target) were 17 ms, compared to 8 ms for humans. Monkey DLs for /pa–ba/ were 25 ms, compared to 11 ms for humans. More pronounced differences between monkeys and humans appeared for prevoiced VOT. Monkey DLs for /ba–mba/ (35 ms) were about four times humans' (8 ms). Monkeys did not reliably discriminate /mba–ba/, although humans experienced no difficulty with this contrast and produced DLs of 18 ms. Monkey VOT sensitivity is compared with that of human infants previously tested with the same stimuli by Aslin *et al.* [Child Dev. **52**, 1135–1145 (1981)]. [Work supported by NIH.]

11:30

O13. Second formant transition discrimination. Joan Besing,[a] John K. Cullen, Robert J. Porter,[b] Randal Rampp, and M. Jane Collins[a] (Kresge Hearing Research Laboratory of the South, LSU Medical Center, New Orleans, LA 70112)

The ability of listeners to process information conveyed by second formant transitions may be expected to be limited by their ability to detect differences in the rate of frequency change for the transition. To examine this question, difference limens (DLs) for initial transitions of isolated second formants were established using a four interval "same–different" procedure. These single formant signals were 300 ms in duration (60-ms transitions and 240-ms steady-state portions). The DLs for starting frequency, and thus rate of frequency change, were established for three standards which varied in rate and direction of transition; all had steady states at 1800 Hz. One standard had no transition and two had transitions of 2.5 Hz/ms that either fell from 1950 Hz or rose from 1650 Hz. The DLs were established for transitions with starting frequencies above and below each standard condition. The DLs for three of four experienced listeners were smaller for transitions in the frequency range above the steady state than for transitions in the frequency range below the steady state. Mean discriminable differences in starting frequency were 67 and 87.5 Hz in the frequency ranges above and below the steady state, respectively. [Supported by NINCDS and the Louisiana Lions Foundation.] [a] Louisiana State University, Baton Rouge, LA 70803. [b] University of New Orleans, New Orleans, LA 70112.

WEDNESDAY MORNING, 13 MAY 1987 REGENCY BALLROOM A & B, 8:30 TO 11:57 A.M.

WEDNESDAY AM

Session P. Speech Communication III: Speech Production

Diane Kewley-Port, Chairman

Department of Speech and Hearing Sciences, Indiana University, Bloomington, Indiana 47405

Chairman's Introduction—8:30

Contributed Papers

8:35

P1. From acoustic tube to acoustic cues. G. M. Kuhn (IDA-CRD, Thanet Road, Princeton, NJ 08540)

The relation between sound propagation in the acoustic tube and formant-based acoustic cues for the phonetic dimension of place of articulation is illustrated. The model for sound propagation comes from Chiba and Kajiyama [*The Vowel* (Phonetic Society of Japan, Tokyo, 1941)]. The evidence for the acoustic cues comes from the experimental literature published after 1941. The following are emphasized: (1) constriction of tubes and changes in formant frequencies; (2) constriction of tubes and invariants in the changes in formant frequencies; and (3) complexities of the changes in formant frequencies and simplifying rules derived from perceptual tests. A tape of modeled speech sounds will be played.

8:47

P2. Articulatory parameters used to generate vocal tract shapes and bark-difference dimensions. Donald C. Wold (Department of Physics and Astronomy, University of Arkansas at Little Rock, 2801 S. University Avenue, Little Rock, AR 72204)

In this study, the correlation between two articulatory parameters used to generate vocal tract shapes from formant frequencies and three bark-difference dimensions were examined. Ladefoged *et al.* [J. Acoust. Soc. Am. **64**, 1027–1035 (1978)] identified the articulatory parameters $w1$ and $w2$ as the front-raising and back-raising proportions of tongue shape and showed how they may be calculated from formant frequencies. The ten American English vowels of Peterson and Barney were used for men and women. The three bark-difference dimensions were described by Syrdal and Gopal [J. Acoust. Soc. Am. **79**, 1086–1100 (1986)]. The $F1$–$F0$ dimension represents vowel height and the $F3$–$F2$ dimension corresponds to vowel place of articulation. The linear correlation between $F1$–$F0$ and $w2$ was -0.86 ($p<0.001$) and that between $F3$–$F2$ and $w1$ was -0.96 ($p<0.001$).

8:59

P3. Difference in second-formant transitions between aspirated and unaspirated stop consonants preceding [ɑ]. Hwei-Bing Lin (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695 and Department of Linguistics, University of Connecticut, Storrs, CT 06268) and Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

Perceptual experiments with synthetic speech have shown that the category boundary on an acoustic [pɑ–tɑ] continuum (obtained by varying the onset frequencies of the second and third formants) is closer to the labial endpoint than the boundary on a [pʰɑ–tʰɑ] continuum. Of several possible explanations, the most plausible seems to be that natural unaspirated and aspirated stops have different formant transitions. To supplement limited data in the literature, an acoustic analysis of CV syllables produced by ten male speakers of American English was conducted. The results show very clearly that the second formants of [pʰɑ] and [tʰɑ] start 100–200 Hz higher than those of [pɑ] and [tɑ], and reach comparable frequency values only at voicing onset. This difference, which is probably an acoustic consequence of subglottal coupling during aspiration, seems to be part of a listener's tacit knowledge of phonetic regularities and thus explains the perceptual boundary shift. It also needs to be taken into account in realistic speech synthesis. [Work supported by NICHD.]

9:11

P4. Observations on the release of /t/. Sandra L. Hamlet (Department of Hearing and Speech Sciences, University of Maryland, College Park, MD 20742)

Palatographic data were examined for the release of prevocalic /t/ in both aspirated and unaspirated contexts. Thirty-two tongue contact points in the alveolar region were sampled every 2 ms. Although a plosive burst is a sudden acoustical event, the dynamic tongue release gesture is more gradual. The burst is associated with the sudden opening of a *narrow central channel* between the tongue and palate. The tongue then continues to pull away from contact centrally, with the final contacts remaining laterally even after voicing for the following vowel has commenced. At some point during the release of /t/, the tongue contact pattern passes through a configuration virtually identical to that for the greatest constriction of /s/. This implies that a brief period of frication generated at the point of release may follow the burst. The sequence of acoustical events at the release of aspirated /t/ would be: burst–frication–aspiration. Tongue releases in a unaspirated context (/st_/) were also examined. The palatographic pattern of tongue release was no different for the unaspirated context, although the timing of its initiation was later relative to the beginning of the following vowel. [Work supported by NIDR.]

9:23

P5. The F1 structure influences final-consonant voicing decisions. Walter V. Summers (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Previous acoustic analyses [W. V. Summers, J. Acoust. Soc. Am. Suppl. 1 **79**, S36 (1986)] showed higher $F1$ steady-state frequencies and higher $F1$ offset frequencies for consonant–vowel–consonant (CVC) utterances ending in voiceless consonants than voiced consonants. These analyses examined the vowels /a/ and /æ/ and the final consonants /b/, /p/, /v/, and /f/. The present experiments examined whether these differences in $F1$ structure provide perceptual information to the listener concerning final-consonant voicing. The CVC stimuli were synthesized with format steady-state frequencies appropriate to /a/ or /æ/ and with final formant transitions appropriate to the bilabial stops, /b/ and /p/. Several series of synthetic stimuli were created differing in $F1$ steady-state frequency, $F1$ final transition slope, and $F1$ offset frequency. Stimuli within a series varied in steady-state vowel duration. At each steady-state duration, stimuli containing higher $F1$ steady-state frequencies were heard as containing voiceless final consonants more often than stimuli containing lower $F1$ steady states. The $F1$ offset frequency also influenced voicing judgments with higher offset frequencies producing more voiceless responses. The $F1$ final transition slope did not have a consistent influence on voicing judgments. [Work supported by NIH.]

9:35

P6. Evidence against acoustic invariance in initial voiced stop consonants. Stephen A. Zahorian, Zaki B. Nossair (Department of Electrical and Computer Engineering, Old Dominion University, Norfolk, VA 23508), and Robert F. Coleman (Eastern Virginia Medical School, P. O. Box 1980, Norfolk, VA 23510)

A perceptual experiment was conducted with naturally spoken intitial stop consonants in order to test recent claims for acoustic invariance in the initial portions of stop consonants [for example, S. E. Blumstein and K. N. Stevens, J. Acoust. Soc. Am. **67**, 648–662 (1980)]. The original stimuli consisted of tokens of the stops /b,d,g/ with the vowels /i,a,u/ as spoken by two male and two female speakers. A computer graphics waveform editor was used to locate the initial portions of the waveform up to the first, second, fourth, and sixth voicing pulses. Three experimental conditions were evaluated: (a) the initial segment only; (b) the initial segment plus the original steady-state vowel; and (c) the intitial segment plus an alternate steady-state vowel. Ten listeners participated in a forced-choice recognition experiment to determine the conditions for which perceptual cues to consonant identity are retained. Recognition accuracy was consistently highest for condition (b), lower for condition (a), and lowest for condition (c). These results, therefore, argue against vowel-independent cues to stop consonant recognition and are more consistent with studies which conclude that the perception of stop consonants is strongly influenced by the following vowel [for example, M. F. Dorman *et al.*, Percept. Psychophys. **22**, 109–122 (1977)]. Thus attempts to isolate the perceptually important cues to stop consonant identity should take into account the vowel context.

9:47

P7. An investigation of some acoustic characteristics of prenasalized stops. Martha W. Burton and Sheila E. Blumstein (Department of Cognitive and Linguistic Sciences, Box 1978, Brown University, Providence, RI 02912)

Prenasalized stops occur in a number of languages, often contrasting phonemically with prevoiced stops, voiceless stops, and nasal consonants. The present study compares and contrasts these sounds in Moru, a Central Sudanic language. A single speaker of Moru produced real word tokens with word-initial prenasalized stops, voiced stops, voiceless stops, and nasal consonants in various vowel environments. Acoustic analyses of these tokens included duration measurements and critical band analyses. The results indicated that prenasalized stops may be characterized by a nasal murmur followed by a 10-dB drop in amplitude, lasting approximately 25–30 ms prior to the release. Critical band analyses of the murmur and prevoiced portions of the prenasalized stops and prevoiced stops, respectively, suggested that the spectral properties differ. In contrast, the spectral properties at the release appear to be similar. These findings will be considered in relation to both phonetic and phonological theories of speech. [Work supported by NIH.]

9:59

P8. "Post-stopped nasals": An acoustic investigation. Marjorie K. M. Chan (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024) and Hongmo Ren (Linguistic Institute, Chinese Academy of Social Sciences, Beijing, People's Republic of China)

In some dialects of Chinese and Miao, the nasals in syllable-initial position have been described as being accompanied by a homorganic stop, which are often transcribed with superscripts: [m^b], [n^d], and [η^g], as a deliberate attempt to characterize these segments as phonetically distinct from prenasalized stops, [$^m b$], [$^n d$], [$^\eta g$]. In our study, the acoustic nature of these "post-stopped" nasals will be explored, since no instrumental study has been conducted on them. A preliminary investigation of data from two Zhongshan Chinese speakers confirm that these nasals are different both perceptually and acoustically from the prenasalized stops in other languages. It is found that the so-called "stop" component in Zhongshan syllable-initial nasals is not a stop, but a burst that occurs simultaneous with the oral release following the nasal. Such bursts occur sporadically in English, but are consistently produced in the Zhongshan nasals, and are perceived as homorganic stops accompanying the nasals. The waveforms also show a characteristic shape, with a sharp rise in amplitude at vowel onset. The results suggest the precise synchronization of velic closure with oral release.

10:11

P9. Pronunciation variation within and across speakers. Michael H. Cohen, Jared Bernstein, and Hy Murveit (Speech Research Program, SRI International, Menlo Park, CA 94025)

An understanding of the structure of pronunciation variation over a population of speakers and over time in the utterances of one speaker should be useful in designing speaker-independent speech recognizers. This paper reports a series of experiments designed to show different kinds of patterns observed in alternative forms of words in constant contexts (e.g., the presence or absence of frication in the "y" in "had your"). An analysis is presented of transcribed data from 630 speakers reading two sample sentences as well as data from four speakers reading the same two sentences 24 times each, separated by filler material, in three separate sessions. The analysis quantifies the relative usefulness of competing models of variation in information theoretic terms. The results indicate that (1) speakers can be clustered into low variation groups such that the variation within a group is significantly less than the population variation, and (2) individual speakers show greater consistency than comparable clustered subsets of the population. Finally, it is suggested how this structure may be used to guide rapid, automatic adaptation in speech recognition. [Work supported by NSE.]

10:23–10:33

Break

10:33

P10. Motor–motor adaptation. Linda I. Shuster (Department of Speech Pathology and Audiology, West Virginia University, Morgantown, WV 26506-6122)

Perceptuomotor adaptation experiments have demonstrated that a purely perceptual task can have a significant effect upon a subsequent production task. Cooper has argued that these perceptual effects upon production reflect a link between the cognitive systems underlying perception and production. This argument might be stronger if it could be demonstrated that the effect that a repetitive perceptual task (such as selective adaptation) has on production is the same as that obtained using a repetitive production task. An experimental technique was developed called motor–motor adaptation. In this task, subjects produced a monosyllabic CV repeatedly (the adaptor). After 20–40 repetitions, the subject produced a second CV syllable once (the test syllable). There were two adaptors ([bi] and [p^hi]) and two test syllables ([di] and [t^hi]). Each test syllable was produced in each adaptor condition. Subjects produced only a single adaptor and a single test syllable within each of the four blocked conditions. Analysis of the obtained voice onset times indicated that for the voiceless test syllable, VOT was significantly shorter after adaptation to [p^hi] than to [bi]. However, for the voiced test syllable, there was no significant difference obtained between the two adaptor conditions. These findings are similar to those Cooper obtained using perceptuomotor adaptation and will be discussed with regard to possible links between speech perception and speech production.

10:45

P11. Articulator movement in anticipatory coarticulation. Gina M. Lee (Department of Linguistics, Ohio State University, 1841 Millikin Road, Columbus, OH 43210)

The results of previous studies examining EMG activity (and, to a smaller extent, articulator movement) in anticipatory gestures have not been consistent. In some studies, the onset of activity for a given feature was *time locked*, in that it began at a fixed interval prior to the acoustic onset of the corresponding segment. This suggests that timing plays an intrinsic role in the organization of speech production, and that segmental targets are dynamic. Other studies have shown evidence for *feature spreading*, where the onset of activity begins within the earliest preceding segment which makes no contradictory demands on the articulators, regardless of the length between the onset of activity and the acoustic onset of the relevant segment. This suggests that time is extrinsic to segmental specification, and that targets are represented as static. However, Gelfer, Bell-Berti, and Harris (1985) and Perkell (1986) argue that the different findings may be due to experimental design: /s/ and /t/, the consonants used in such studies, may not be unspecified with regard to lip protrusion, as previously assumed. In the current study, the issue of feature specification for presumed neutral segments is considered. Pellet traces from selected bases of x-ray microbeam data were examined. The point of onset of lip protrusion with respect to the acoustic onset of the rounded vowel in VC_nu utterances was measured. Velocity patterns of articulator movement were also considered.

10:57

P12. Interarticulatory timing and single articulator velocity-displacement relationships in English stress pairs. Kenneth deJong (Department of Linguistics, Ohio State University, Columbus, OH 43210)

Evidence has been brought forward that there exists a direct timing relationship between the articulator movements associated with vowel production and those associated with consonant production. Tuller and others have interpreted such results as evidence for a model of speech production that posits a metrical relationship between vowels and their coproduced consonantal neighbors [cf. Harris *et al.*, in *Invariance and Invariability in Speech Production*, edited by J. Perkell and D. Klatt (Erlbaum, Hilldale, NJ, 1986); and Tuller *et al.*, J. Exp. Psychol. **9**, 829–833 (1983)]. A more strictly segmental ordering of speech events could be posited to handle such data. The difference between these models can be brought to light by making reference to statistical correlations between the timing latencies of various articulatory events [cf. K. Munhall, J. Acoust. Soc. Am. **78**, 1548–1553 (1985)]. This study investigates the effects of stress pattern and syllable structure on articulator timing, using x-ray microbeam traces of jaw, tongue blade, and lip movement. Unlike in earlier studies, real English words were used. Each token was placed in two contexts, natural sentences and short, frame sentences. The effects of these two environments are also to be noted. The traces will also be used to replicate the velocity-displacement relationship in the movement of the jaw, found in earlier studies of reiterant speech [Kelso *et al.*, J. Acoust. Soc. Am. **77**, 266–280 (1985)].

11:09

P13. Vocal fold contact area. Ingo R. Titze (Department of Speech Pathology and Audiology, The University of Iowa, Iowa City, IA, 52242 and the Recording and Research Center, Denver Center for the Performing Arts, 1245 Champa Street, Denver, CO 80204), David Druker, and Paul Durham (Department of Speech Pathology and Audiology, The University of Iowa, Iowa City, IA 52242)

Vocal fold contact area is generally inferred from the electroglottographic (EGG) signal. To our knowledge, only one attempt has been reported to measure contact area directly [Scherer, Druker, and Titze, Vocal Fold Physiology Conference, Tokyo, Japan (1987)]. This direct measurement is needed to identify artifacts and ambiguities in the EGG signal that may lead to questionable interpretations of vocal fold movement. In particular, the spatial distributions of contact area (anterior–posterior and vertical) is not always clear by simple inspection of the EGG signal. In this study, excised dog larynges were prepared and mounted such that one vocal fold could vibrate against a sheet of electroconductive glass. The area of contact was then extracted by videostroboscopy and image processing techniques. Comparisons are made between three-parameter and four-parameter simulation models of EGG and the experimentally determined contact area. [Work supported by NINCDS.]

11:21

P14. Source-system coupling in voice production in excised larynges. Donald S. Cooper and Steven H. Florman (Department of Otolaryngology, University of Southern California, Hoffman 903, 2025 Zonal Avenue, Los Angeles, CA 90033)

An apparatus was constructed to study the effects of subglottal and supraglottal resonances on voice production in excised human larynges. The approach was based on an earlier study of W. Trendelenburg and H. Wullstein [summary by D. S. Cooper in *Otolaryngology—Head and Neck Surgery*, edited by C. W. Cummings *et al.* (Mosby, 1986), Vol. 3, pp. 1766–1776]. The resonance of the pseudolung was adjustable, while variation of supraglottal resonances was achieved by construction of vowel resonators based on Fant's area functions. Activity of intrinsic laryngeal muscles was simulated by weights attached through pulleys to the larynx. In order to examine the effect of acoustic loading with a full range of variation of the first formant, voice production with supraglottal resonances based on /i/ and /ɑ/ was compared with phonation without the resonator. The resting glottal configuration and average air flow were maintained constant, while the effects of varying supraglottal resonances on voice production were observed in terms of acoustic and glottographic measurements.

11:33

P15. The stability of phonational frequency range when elicited by two different methods. Connie Cook Spencer and Marylou Pausewang Gelfer (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Previous investigations of phonational frequency range suggest that the method of elicitation can affect the stability of this range. The method of matching tones [R. Colton and H. Hollien, J. Speech Hear. Res. **15**, 708–713 (1972)] appeared to yield more stable data than did spontaneously produced scales [X. Gelfer, J. Acoust. Soc. Am. Suppl. 1 **79**, S82 (1986)]. In the present study, subjects matched descending and ascending semitones and spontaneously sang down and up the scale to the lower and upper limits of their range. These data were collected three times in one day, and three times on a second day 10–12 weeks later. The elicited frequency productions were converted to semitones (ST), and lowest and highest production and total range were calculated. For both elicitation methods, within-day and between-day variability were examined for all measures. Results indicate that similar frequency range values are obtained regardless of elicitation method. However, for both lower and upper limit of range, the matching procedure yielded somewhat more extreme values (0.2 ST, 0.4 ST). Within-day and between-day variation were similar to what has previously been reported (± 2 ST).

11:45

P16. The perception of voicing in word-final Catalan stops. Jan Charles-Luce (Department of Linguistics, Indiana University, Bloomington, IN 47405)

It has been demonstrated previously that Catalan does not always exhibit complete acoustic neutralization of the underlying voice contrast in word-final stops. The present study investigated whether listeners can use the acoustic differences distinguishing underlying voicing to identify the voicing of work-final stops. Native Catalan and English speakers were auditorily presented with natural tokens of three Catalan minimal pairs differing in the underlying voicing of the word-final stop. For these minimal pairs, vowel duration significantly distinguished underlying voicing in two-sentence medial position, but not in a sentence final position. The minimal pairs were excised from the sentences and presented in isolation. The results showed that neither the Catalan nor the English listeners were able to identify significantly above chance the underlying voicing of word-final stops produced in any of the sentential positions. Thus, the acoustic differences differentiating the underlying voice contrast do not appear to be perceptually functional for distinguishing minimal pairs. The apparent dichotomy of production and preception grammars and the implications for phonological neutralization will be discussed. [Work supported by NIH Training Grant NS-07134.]

Session Q. Underwater Acoustics III: Numerical Solutions of Two Benchmark Problems

Leopold B. Felsen, Chairman

Department of EECS, Polytechnic University, Farmingdale, New York 11735

Chairman's Introduction—8:30

The First Special Session on Accuracy Assessment of Numerical Codes was held at the 112th Meeting of the Acoustical Society in Anaheim, California, 8–12 December 1986. An outcome of that session was a consensus to pose specific benchmark problems for numerical implementation. Accordingly, two range-dependent propagation problems have been posed and distributed to the underwater acoustics community, and contributions have been solicited for presentation at this meeting. Because the outcome of the calculations could not be predicted at the submission deadline for abstracts, the traditional procedure has been modified to require potential serious contributors to submit only their name(s) and affiliation(s), with a statement of which problem(s) they plan to attack and what code(s) they intend to apply. Presentors are expected to discuss how well their code is suited to the problem at hand, what accuracy controls were imposed, what computational difficulties were encountered, what computation time was required, etc. It has been requested that data are plotted in a standardized format so as to facilitate direct comparison between results produced by different algorithms. Problem I deals with upslope propagation in a wedge-shaped channel while Problem II models a plane parallel waveguide with range-dependent sound-speed profile.

Problem I—Coordinators and Discussion Leaders:

M. Buckingham
Massachusetts Institute of Technology, 5-204, Cambridge, Massachusetts 02139

F. Jensen
Group Leader, Environmental Modeling, NATO, SACLANT Research Centre, APO New York, New York 09019

R. Stephen
Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

Problem II—Coordinators and Discussion Leaders:

J. DeSanto
Center of Wave Phenomena, Department of Mathematics, Colorado School of Mines, Golden, Colorado 80401

D. Wood
Naval Underwater Systems Center, Code 3122, New London, Connecticut 06320

Contributors:

(Presentation order and time to be determined by coordinators.)

Q1. Analytic solutions to three wedge problems. Michael J. Buckingham (Massachusetts Institute of Technology, Cambridge, MA 02139) and Alexandra Tolstoy (Naval Research Laboratory, Washington, DC 20375)

Q2. Parabolic equation and coupled mode solutions. Alastair Cowley and Roy G. Levers (Admiralty Research Establishment, Portland, Dorset DT5 2JS, England)

Q3. F. E. parabolic equation code. Michael D. Collins (Northwestern University, Evanston, IL 60201)

Q4. Paraxial ray solutions. Gregory L. Duckworth, Jung M. Lee, and Roger Turpening (Earth Resource Laboratory, Massachusetts Institute of Technology, Cambridge, MA 02139)

Q5. Convergence rate of Gaussian quadrature for classical ray tracing. Edward R. Floyd (Arctic Submarine Laboratory, Naval Oceans System Center, San Diego, CA 92152)

Q6. Model experiments for 3-D propagation characteristics in a wedge-shaped ocean. Stewart A. L. Glegg and Jong R. Yoon (Florida Atlantic University, Boca Raton, FL 33431)

Q7. Coupled mode and parabolic equation solutions. Finn B. Jensen (SACLANT ASW Research Centre, 19026, La Spezia, Italy)

Q8. Split step Fourier algorithm with high angle approximation. Lan Nghiem-Phu (Daubin Systems Corporation, Miami, FL 33149)

Q9. Parabolic equation implicit finite difference (IFD) solutions. J. S. Robertson, D. C. Arney (U.S. Military Academy, West Point, NY 10096) W. L. Siegman and M. J. Jacobson (Rensselaer Polytechnic Institute, Troy, NY 12180)

Q10. Finite difference solutions. Ralph A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

Q11. Wide angle parabolic equation solutions. David Thomson (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V05 1B0, Canada)

Q12. Numerical implementation of DeSanto's parallel waveguide problem. David Thomson, Gary Brooke (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V05 1B0, Canada), and John DeSanto (Colorado School of Mines, Center for Wave Phenomena, Golden, CO 80401)

Other Contributions:

Because of the short time between the benchmark problem statement and the abstract due date, a few additional contributions will be accepted. Please contact Prof. L. B. Felsen.

Chairman's Summary: **(Time determined by chairman and session coordinators.)**

WEDNESDAY MORNING, 13 MAY 1987 MT. RAINIER, 9:00 TO 11:35 A.M.

Session R. Noise III and Shock and Vibration I: Instrumentation Calibration—Facts and Fallacies

John P. Seiler, Chairman

Mine Safety & Health Administration, U. S. Department of Labor, 4800 Forbes Avenue, Pittsburgh, Pennsylvania 15213

Chairman's Introduction—9:00

Invited Papers

9:05

R1. Instrument calibration: A legal perspective in the context of enforcement. Manuel R. Lopez (Office of the Solicitor, U. S. Department of Labor, 4015 Wilson Boulevard, Arlington, VA 22203)

There is widespread use of noise dosimeters, sound level meters, and acoustical calibrators in statutory enforcement schemes. As a general rule, there are three prerequisites to be met in order for evidence obtained through the use of these instruments to be sufficient and admissible. First, in the performance of the intended function, the scientific reliability of the instrument must be established. This is often accomplished by showing that the instrument is permitted by law or acknowledged as reliable by the court. Second, it must be established that the instrument has been properly operated. This is done by showing that the operator is properly trained, followed applicable instructions, and is familiar with the instrument. Third, the accurate calibration of a particular instrument that is used in a given case must be shown; statutory calibration procedures, where they exist, must be strictly adhered to; acoustical calibrators where used to calibrate other instruments must also be calibrated; and, should the accuracy of the instrument to be calibrated contain a variance, enforcement allowances should be made.

9:35

R2. Traceability of acoustical instrument calibration to reciprocity calibrations of laboratory standard microphones at the National Bureau of Standards. Victor Nedzelnitsky (National Bureau of Standards, Building 233 (Sound), Room A149, Gaithersburg, MD 20899)

Critical needs of public and private acoustical calibration laboratories for traceability to a reliable source have led to measurement services based upon the pressure and/or free-field calibration of laboratory standard microphones by established reciprocity methods over wide frequency ranges at the NBS. These primary methods and measurement services provided by the NBS at fixed cost to other laboratories are described, and the frequency-dependent uncertainties associated with these methods and services are discussed. Instrument calibration hierarchies comprising direct or implied chains of traceability to the NBS may involve at least four different definitions of traceability, and one or more of a variety of secondary calibration methods. Assessing the adequacy of a particular system for realizing traceability involves judging the capacity of that system to ensure measurements of acceptable accuracy for their intended purpose. Examples of traceability and the frequency-dependent uncertainties in some methods used to achieve it are discussed.

10:05

R3. Calibration capabilities—Army Primary Standards Laboratory. James R. Arrington (U. S. Army TMDE Support Group, ATTN: AMXTM-SP, Redstone Arsenal, AL 35898-5400)

The Army Primary Standards Laboratory (APSL) generates and maintains measurement standards in acoustics and vibration. Army measurement activities are supported by a three-level system that provides increasing degrees of accuracy starting with field calibration teams and ending with APSL. Traceability, calibration methods, calibration accuracy, transducer densities, and transducer types are discussed.

10:35

R4. Laboratory calibration of acoustical calibrators and noise dosimeters. Michael A. Crivaro and Dennis A. Giardino (U. S. Department of Labor, MSHA, 4800 Forbes Avenue, Pittsburgh, PA 15213)

The Mine Safety and Health Administration (MSHA) maintains a laboratory to calibrate noise dosimeters and acoustical calibrators. Dosimeters and calibrators are used by Federal mine inspectors to determine compliance with occupational noise regulations. Topics covered include reasons for creating the laboratory, test system design philosophy, test procedures, and estimates of uncertainties.

11:05

R5. The calibration of integrating-averaging sound level meters: A manufacturers point of view. Peter Hedegaard (Brüel & Kjær Industri A/S, Nærum, Denmark)

The introduction of the integrating-averaging SLM into common use has given rise to some new calibration and testing problems compared with those already known from the ordinary SLM. Determination of reasonable requirements for the integrating-averaging section and how to verify the performance has been a matter of discussion in connection with the national and international standardization work over the past years. Some aspects concerning the requirements and the test methologies and philosophies are discussed. The new opportunity to carry out spatial averaging in a simple way has given new life to the old discussion about free-field and random-field calibration. The free-field/random-field philosophies are summarized and some ideas about the possibilities for a final closing of "the Atlantic divide" are given.

Session S. Physical Acoustics III: Scattering II: Waveguides, Imaging, Flow, and Probes

Joseph A. Clark, Chairman
Mechanical Engineering Department, Catholic University, Washington, DC 20064

Contributed Papers

9:00

S1. Comparison of backscattered and forward scattered fields for objects immersed in a shallow water waveguide. M. F. Werby and Guy Norton (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL, MS 39529)

A noise source will form a guided wave in a suitable environment at sufficiently high frequencies. When the guided wave encounters a submerged object, the field scattered from the object will itself form a guided wave when suitably far from the object. The object scatters the field in a fairly complicated manner and depends on the relative direction of measurement from that of the incident field as well as the orientation of the object relative to the guided wave. A recent numerical development at NORDA enables calculations to be simulated for such physical situations. The objective of this study is to assess the relative importance of forward scattered to backscattered fields. Calculations are performed and compared for several frequencies for the two configurations and will be presented. Results indicate that, above a frequency limit, forward scatter is stronger, while, below that frequency, backscatter is favored.

9:15

S2. Acoustic scattering in an ocean environment: III. Scattering in an inhomogeneous layered waveguide. Roger H. Hackman and Gary S. Sammelmann (Naval Coastal Systems Center, Panama City, FL 32407)

Previously, the formal solution to the scattering from an elastic target in a waveguide with an arbitrary number of homogeneous layers was presented [R. H. Hackman and G. S. Sammelmann, J. Acoust. Soc. Am. Suppl. 1 **78**, S76 (1985)]. The solution is valid to all orders of multiple scattering among the target and waveguide boundaries. Here, this scattering formalism is extended to an inhomogeneous layered waveguide with an arbitrary number of layers. As a first application of this formalism, the long-range, low-frequency scattering from an elastic spherical shell in a range-independent waveguide with a sound-speed profile of the kind leading to caustic formation is considered. The focus of this study is the extent to which the elastic information in the scattered wave is modified by the propagation of both the incident and scattered signals through a convergence zone. [Work supported by the Office of Naval Research.]

9:30

S3. Acoustic scattering in a range-independent, shallow water waveguide with a penetrable bottom. Gary S. Sammelmann, D. H. Trivett, and Roger H. Hackman (Naval Coastal Systems Center, Panama City, FL 32407)

At a previous meeting of the Acoustical Society of America, the analysis of the low-frequency, acoustic scattering from an elastic spherical shell in a homogeneous, range-independent waveguide with an impenetrable bottom was presented [G. S. Sammelmann and R. H. Hackman, J. Acoust. Soc. Am. Suppl. 1 **79**, S76 (1986)]. This previous investigation centered on elucidating the underlying dynamical picture leading to the observed "fine structure" of the scattering resonances of the sphere in the waveguide and to the occurrence of "superresonances." Here, this analysis was extended to a waveguide with a (liquid) penetrable bottom to more completely explore the dependence of the resonance spectrum of the sphere on the acoustic environment. For a sphere in the sediment layer, the effects of the sediment loading and of the partial "unloading" by the nearby sediment–seawater boundary are discussed. [Work supported, in part, by the Office of Naval Research.]

9:45

S4. Scattering of acoustic waves in a waveguide. Rahul Sen (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24060) and Charles Thompson (Department of Electrical Engineering, University of Lowell, One University Avenue, Lowell, MA 01854)

The problem of scattering from boundary discontinuity in a waveguide is discussed. The relationship between the static and dynamic representations of the scattered pressure field will be investigated for those frequencies falling below the first cross mode of the duct. Special attention is paid to the influence of cutoff cross modes to the solution of the pressure field. It is shown that the method of matched asymptotic expansions can be successfully used to determine globally valid pressure field junction conditions near a boundary discontinuity. The matching of exponentially decaying terms of the inner solution is shown to, in turn, contribute to the junction impedance and extend the frequency range of the solution's validity.

10:00

S5. Experiments on junction inertance. Zili Li and A. H. Benade (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

The concept of junction inertance at the joint between two axisymmetric waveguides of different tapers arises from the nonaxial rearrangement flow and may be understood in terms of the imaginary wave impedance of the locally excited evanescent modes. The cases considered consist of cylindrical pipes with infinitely long conical terminations of different angles. In these cases, the terminal impedance of the pipe can be written as $Z_t = j\omega M_j + Z_{\text{cone}}$, where M_j is the junction inertance and Z_{cone} is the parallel combination of M_c, conicity inertance of the cone, and R_0, the characteristic impedance of the pipe. Experiments were performed over a frequency range of 150 Hz–15 kHz and reflection coefficients were measured for cones with half-angle 10, 32, and 90 deg. The results showed that at high frequencies (above 8 kHz for pipe radius $a = 10$ mm), the effect is negligible for all cases. For 10 deg, M_j is $0.03(a/c)R_0$ in a range of 4–6 kHz, where c is the speed of sound. At low frequencies (below 3 kHz), Z_t has the form of $j\omega(M_j + M_c)$; to separate M_j and M_c becomes more complicated. Several cases will be discussed. [Work supported by NSF.]

10:15

S6. Analytic solutions to the wave equation in a duct lined with axially nonuniform absorbents. P. G. Vaidya and D. G. Martin (Washington State University, Pullman, WA 99164-2920)

For circular ducts with axially uniform boundary conditions, the acoustic field can be expressed in terms of a set of eigenfunctions. The corresponding eigenfunctions depend on the circumferential and the axial mode number. In this paper, a modification of the analysis, when the axial boundary conditions are nonuniform, is presented. The first part of the paper deals with an exponential variation in a semi-infinite duct. It has been shown that the field can be represented in terms of submodes. The eigenvalues of each of these submodes are distinct from each other, however, if they can be readily calculated. This analysis then is extended to an arbitrary variation of axial boundary conditions for finite length ducts. Numerical calculations are presented for the special case of circumferentially uniform fields ($m = 0$). It has been noticed that the character of the multiple eigenvalues of the submodes undergoes a qualitative change (akin to nonlinear bifurcation phenomenon) at the third radial mode ($n = 3$), for the specific case studied.

10:30

S7. Analytical solutions to the wave equation in a duct lined with circumferentially nonuniform absorbents. D. G. Martin and P. G. Vaidya (Washington State University, Pullman, WA 99164-2920)

Nonuniform boundary conditions are of fundamental interest from the theoretical point of view. They are also important from a practical point of view because of the potential of substantial enhancement of attenuation, which has been experimentally demonstrated. However, conventional eigenfunction solutions cannot be used to solve such problems. Therefore, much of the work in this area has been of a purely numerical nature. In this paper, the recent work by Vaidya, which was restricted to a single term in the exponential expansion, has been extended to any arbitrary circumferential variation. It has been found in this general case that the concept of submodes can still be used to construct the solutions. However, the eigenvalues of submodes can be altered, as opposed to the observation in the restricted case. A modified procedure to find the eigenvalues and the submodal coefficients has been developed. Specific numerical calculations are presented to show the alteration of the acoustic field and attenuation in hollow circular ducts.

10:45

S8. Modal propagation in ducts lined with elastic porous materials. J. S. Bolton and N. M. Shiau (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

The characteristics of sound propagation in ducts lined with either locally or bulk reacting material are well known. The latter material is generally assumed to be homogeneous and capable of supporting only a single longitudinal wave and is usually modeled as a rigid-framed porous material. It has recently been demonstrated both experimentally and theoretically that elastic porous materials of the type used in noise control (i.e., relatively stiff, partially reticulated foam) can support two longitudinal waves: the frame wave and the airborne wave. In this paper, Scott's treatment of extended reaction duct linings is modified to allow for this observation. Equations governing the three-dimensional propagation of each wave type within the material are presented along with the appropriate boundary conditions at the duct/lining interface. The effects of finite frame elasticity are demonstrated by a comparison of the attenuation and phase speed of the first several duct modes for elastic porous linings and the equivalent rigid porous linings.

11:00

S9. Bragg imaging of high-intensity ultrasonic waves. M. A. Breazeale, Jeong Kwan Na (Department of Physics, University of Tennessee, Knoxville, TN 37996-1200), and Oswald Leroy (Katholieke Universiteit Leuven, B-8500 Kortrijk, Belgium)

Light incident on ultrasonic waves at the Bragg angle produces a diffraction pattern described by $n\lambda = 2\lambda * \sin\theta_B$. Korpel [*Optical Imaging of Ultrasonic Fields by Acoustic Bragg Diffraction* (Rotterdam, 1962)] studied, in detail, the condition $n = 1$ and the image produced in the first diffraction order. Martin [J. Appl. Phys. **43**, 1480 (1972)] observed Bragg diffraction with high-intensity ultrasonic waves and noted that for $n \geqslant 2$ one observes multiple images in the diffraction orders. A somewhat successful modification of the mapping theory of Korpel was developed, although it had distinct limitations. The present discussion is focused on $n \geqslant 2$ and the optical conditions leading to multiple images in the higher diffraction orders. The existence of two images is conveniently explained on the basis of the theory of Blomme and Leroy [Acustica **59**, 1821 (1986)]. [Research supported by the U.S. Office of Naval Research and by the UT-ORNL Science Alliance.]

11:15

S10. Ultrasonic Bragg imaging of flaws. Jeong Kwan Na, M. A. Breazeale (Department of Physics, University of Tennessee, Knoxville, TN 37996-1200), and Oswald Leroy (Katholieke Universiteit Leuven, B-8500 Kortrijk, Belgium)

A superficial crack on an aluminum plate becomes an internal flaw when covered by a second plate. Ultrasonic waves are caused to resonate between the external surfaces of the plates and produce transmission maxima characteristic of Lamb modes in the plates. The transmitted wave fronts in a liquid, modified by the presence of the flaw, are now used to diffract light at the Bragg angle. The Bragg diffraction orders contain images of the ultrasonic wave fronts and, hence, can be used to image the internal flaws. Characteristic flaws are shown and resolution limits are discussed. [Research supported by the U.S. Office of Naval Research and by the UT-ORNL Science Alliance.]

11:30

S11. Accurate and precise assessment of pulsatile volume blood flow by time domain correlation. Ilmar A. Hein and William D. O'Brien, Jr. (Department of Electrical and Computer Engineering, University of Illinois, 1406 W. Green Street, Urbana, IL 61801)

An ultrasonic flow velocity measurement technique using time domain correlation has been developed to accurately and precisely measure the axial flow velocity of liquid in a circular vessel. This technique can be used to estimate the volume flow in a circular vessel without any previous knowledge of the vessel size, flow velocity profile, or transducer measurement angle. Continuous flow has been measured with an overall uncertainty of less than 15%. Pulsatile flow has been estimated by measuring flow velocity at the center of the vessel. Results indicate that the accurate and precise measurement of pulsatile flow of human blood by time domain correlation is possible, and the application of this technique in the diagnosis of various thrombosis in the lower limbs is being studied.

11:45

S12. Measurement of coherence with a vector acoustic intensity probe. Joseph A. Clark (Mechanical Engineering Department, Catholic University, Washington, DC 20064 and DTNSRDC, Bethesda, MD 20084)

Complex coherence is usually defined in the acoustics literature as the (ensemble-averaged) cross spectrum between pressure measurements made simultaneously at two points and normalized with respect to the square root of the product of the autospectra at the same two points. If the two points are sufficiently close together, a considerable simplification occurs which allows the complex coherence to be modeled as a function of one position with two parts. The real part corresponds to the classical definition of plane-wave intensity. However, it is scaler in character (omnidirectional). The imaginary part corresponds to the definition of intensity assumed by acoustic intensity measurement methods of more recent interest. This part exhibits a vector character. Vector acoustic intensity probes consisting of at least four elements arranged in closely spaced pairs along three orthogonal directions, can completely characterize complex coherence fields of the form described above. Ambient noise fields (and fields containing both signals and noise) can, in turn, be characterized by complex coherence measurements. These characterizations are of practical importance in estimating the accuracy and array gain of acoustic intensity measurements.

Session T. Architectural Acoustics III, Psychological and Physiological Acoustics V, and Speech Communication IV: Speech Intelligibility in Lecture Halls and Classrooms

Richard M. Guernsey, Chairman
Cedar Knolls Acoustical Laboratories, 9 Saddle Road, Cedar Knolls, New Jersey 07927

Chairman's Introduction—9:30

Invited Papers

9:35

T1. Review of methods of measuring rooms for speech. J. S. Bradley (Institute for Research in Construction, National Research Council of Canada, Ottawa K1A 0R6, Canada)

Measures of the acoustical quality of rooms for speech should include both the signal-to-noise and the room acoustics aspects of the overall problem. The classical approach was to include these two aspects separately as background noise levels and reverberation times. Newer measures have combined both aspects into a single quantity. These more successful newer measures include an assessment of the beneficial effects of the direct and early reflected sounds and the detrimental effects of later arriving sounds. The prediction accuracies of a number of acoustical measures are compared, and optimum values of each quantity are presented. Optimum values for the classical quantities, background noise levels, and reverberation times are derived as a function of room volume from the newer quantities. The relationships between various measures are discussed.

10:00

T2. Speech intelligibility in classrooms using RASTI. Klaus Hojbjerg (Brüel & Kjaer Instruments, 185 Forest Street, Marlboro, MA 01752)

Speech intelligibility in classrooms is normally not a problem. However, in certain situations, when teaching hearing-impaired children, the acoustics of the classroom becomes important. Many old classrooms have insufficient acoustic design and therefore must be treated when hearing-impaired children are integrated with children with normal hearing. A quick method of assessing speech intelligibility is the RASTI system. This method, developed by Houtgast and Steeneken in The Netherlands, is based on a measure of the modulation transfer function in two octave bands. These two bands are centered at 500 Hz and 2 kHz, where a total of nine modulation frequencies are measured. This paper will describe the RASTI system, and actual measurements will be shown.

10:25

T3. Standardization of speech level measurement. Karl S. Pearsons (BBN Laboratories, Inc., 21120 Vanowen Street, Canoga Park, CA 91303)

The level of speech is measured using a variety of instrumentation and techniques. Most methods serve the needs of the particular user. However, the many different schemes make it difficult to compare one set of speech levels with another. In an attempt to provide a standard method of measurement, a standards working group was formed to develop a simple, meaningful, repeatable method for quantifying speech. This paper discusses the difficulties of providing such a measure and presents the philosophy of the working group's proposed method. The proposed method will be described and relationships with other speech measurement techniques will be presented.

10:50

T4. Speech communication in lecture halls and classrooms for hearing-impaired. Anna K. Nabelek (Department of Audiology and Speech Pathology, The University of Tennessee, Knoxville, TN 37996-0740)

Rooms with good acoustics for normal-hearing listeners might be inadequate for people with hearing impairments. To achieve listening conditions satisfactory for the hearing-impaired listeners, three conditions must be met: (1) Speech should be delivered at higher sound-pressure level than for normal-hearing listeners to compensate for hearing loss; (2) background noise should be at least 15 dB lower than the level of speech; and (3) the reverberation time of the room should be as short as possible. Amplification through public address systems is not practical because hearing-impaired listeners would need levels not tolerated by the general public. Also, because there are limits to reduction in background noise and elimination of sound reflections, listening conditions in lecture halls and in classrooms might be less than ideal for the hearing-impaired. A solution is the use of assistive listening systems. In these systems, sounds which are picked up by microphones located close to the talkers' mouths contain little noise and reverberation. Then the sounds are transmitted on radio or infrared waves throughout a whole room. The sounds are received by individually worn receivers which are equipped with gain controls. The benefits and limitations of assistive listening systems will be discussed. [Work supported by NIH.]

Contributed Paper

11:15

T5. Speech intelligibility measurements using TEF analysis. Don Davis (Synergetic Audio Concepts, P.O. Box 1239, Bedford, IN 47421), Don Keele (Crown International, 1718 W. Mishawaka Road, Elkhart, IN 46517), and Eugene Patronis (1774 Northridge Road, Dunwoody, GA 30338)

Recent investigations into the measurement of speech intelligibility using large listener groups (three groups of 30 each) in a cathedral, a concert hall, and a classic motion picture theater of the early 1930s suggest that a listener's subjective intelligibility score will be best matched in objective testing by dividing the sound considered to be direct sound from sound considered to be reverberant sound at the highest level return within the first 50 ms. This, in a majority of cases, results in using only the first arrival and no integration of early sound. Evidence will be presented that indicates that the use of impulse squared measurements that fail to process the signal as a complex analytic signal often fail to properly record major reflective signals unless extensive spatial averaging is resorted to. The paper will be supported by actual postprocessing of data taken with a Techron TEF analyzer using the Heyser technique for energy time curve measurement. In such measurements, minor movements of the microphone dramatically affect the impulse and doublet responses while the same change barely affects the energy time curve measurement, thus demonstrating the need for the complex analytic signal in such analysis.

11:30–12:00

Panel Discussion

Session U. Bioresponse to Vibration I: Vibratory Sensitivity and Tactile Speech Aids

Janet M. Weisenberger, Chairman
Central Institute for the Deaf, 818 S. Euclid, St. Louis, Missouri 63110

Contributed Papers

1:00

U1. Comparison of two multichannel tactile aids for the hearing-impaired. Susan M. Potts and Janet M. Weisenberger (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Two multichannel tactile devices for the hearing-impaired were compared in tasks using acoustic stimuli of varying levels of complexity. One device was a 16-element linear vibratory array worn on the forearm, which displayed activity in 16 overlapping frequency channels. The other device utilized 16 channels in a similar stimulus processing strategy, but delivered the tactile stimulation to a linear electrocutaneous array worn on the abdomen. Subjects were tested in tasks including phoneme discrimination in pairs of rhyming syllables, phoneme identification in larger sets of rhyming syllables, learning of a list of words, and connected discourse tracking. All stimuli were presented live-voice by a female talker, and lipreading was permitted only in the tracking task. Results showed both devices to be good transmitters of manner and voicing features of articulation, but poor transmitters of place features. Differences between the two devices were found only at the level of connected discourse tracking. Results are discussed in terms of differences between the two devices in processing strategy, location of stimulator array, and type of transducer. [Work supported by NSF and NIH.]

1:15

U2. Vibrotactile masking as a function of stimulus-onset-asynchrony. G. A. Gescheider (Psychology Department, Hamilton College, Clinton, NY 13323), R. T. Verrillo (Institute for Sensory Research, Syracuse University, Syracuse, NY 13210), and S. J. Bolanowski, Jr. (Center for Brain Research, University of Rochester Medical School, Rochester, NY 14642)

Vibrotactile thresholds for the detection of a 50-ms vibratory stimulus on the thenar eminence of the hand were measured in the presence of and in the absence of a 700-ms suprathreshold vibratory masking stimulus. When thresholds were measured in the presence of the masking stimulus, stimulus-onset-asynchrony was varied so that backward, simultaneous, and forward masking could be measured. The amount of masking, expressed as the difference between thresholds for detecting the test stimulus in the presence of and in the absence of the masking stimulus, was greatest when the test stimulus was presented near the onset or offset of the masking stimulus. For both backward and forward masking, the amount of masking decreased as a function of increasing stimulus-onset-asynchrony with the decay rate being greater for backward than for forward masking. Comparisons were made of the amounts of masking measured when the test and masking stimuli were both sinusoids, when the test and masking stimuli were both noise, and when the test stimulus was a sinusoid and the masking stimulus was noise.

1:30

U3. Psychophysical evidence for a four-channel model of vibrotaction. Stanley J. Bolanowski, Jr. (Center for Brain Research, University of Rochester Medical School, Rochester, NY 14642), George A. Gescheider (Department of Psychology, Hamilton College, Clinton, NY 13323), Ronald T. Verrillo (Institute for Sensory Research, Syracuse University, Syracuse, NY 13210), and Christine M. Checkosky (Center for Brain Research, University of Rochester Medical School, Rochester, NY 14642)

Although previous physiological experiments have identified four sensory channels (Pacinian, RA I, SA I, and SA II) in glabrous skin of the human somatosensory periphery, only three (Pacinian, RA I, and SA I) have been shown to mediate vibrotactile sensation. By using stimuli to selectively mask these various channels, it is demonstrated that a fourth channel does, indeed, participate in the perceptual process. The fourth channel is unaffected by changes in skin-surface temperature (15° to 40 °C) and stimulus area (0.008 to 2.9 cm^2) or duration (700 to 2500 ms). Adhering to previous psychophysical nomenclature, the fourth channel is called non-Pacinian III (NP III). The four psychophysically measured channels (P, NP I, NP II, and NP III) combine at threshold to create an operating range for the perception of vibration which extends from 0.4 to >500 Hz. The four channels mediate specific portions of the overall threshold-frequency characteristic, although their sensitivities partially overlap. Thus suprathreshold stimuli may activate several channels simultaneously, suggesting that the perceptual qualities of touch may be determined by the combined inputs of all four channels. [Work supported by NIH and NSF.]

1:45

U4. Lingual vibrotactile magnitude estimation and cross-modality matching: Aging effects. Elizabeth Randolph-Tyler (Ohio University, Athens, OH 45701), Donald Fucci (Ohio University, Athens, OH 45701), Linda Petrosino (Bowling Green State University, Bowling Green, OH 43403), and Daniel Harris (Healthcare Rehabilitation Center, Austin, TX 78745)

Lingual vibrotactile suprathreshold sensation magnitudes were investigated across four age groups (mean ages = 7.8, 19.3, 45.2, and 57.0 years) by employing the psychophysical methods of magnitude estimation and cross-modality matching. Lingual vibrotactile stimuli were presented in combination with auditory stimuli for the cross-modality matching task. For lingual vibrotactile magnitude estimation, both the upper and lower power functions were steeper for the oldest age group. The power functions for cross-modality matching in which the vibratory stimulus was the standard showed asymptotic growth at about 25-dB sensation level for the three older age groups, but not for the youngest age group. Straight-line power functions were obtained for all age groups on the cross-modality matching task when the auditory stimulus was the standard, with the older aged subjects making larger lingual vibrotactile magnitude adjustments to the lower level auditory stimuli than the younger aged subjects. In summary, age-related response differences did

occur at suprathreshold levels of vibrotactile stimulation, and the psychophysical techniques employed appeared to be a noninvasive way to make judgments about suprathreshold functioning of the human tactile system as it changes with age.

2:00

U5. Tactile sensation in hands exposed to vibration: A pilot study. A. J. Brammer, J. E. Piercy (Division of Physics, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada), P. L. Auger (Centre Hopitalier Université Laval, Ste. Foy, Québec G1V 2K8, Canada), and S. Nohara (Department of Public Health, School of Medicine, Kanazawa University, Kanazawa, 920 Japan)

Three methods are compared for assessing impaired tactile sensation in vibration-exposed workers: a medical examination including traditional neurological tests, and refined measures of vibrotactile and gap perception. Of 28 subjects, 21 were judged free of confounding factors (13 forestry workers exposed to chain saw vibration, aged 28 years $\pm$ 4 s.d., and 8 laboratory workers not exposed to vibration, for comparison, aged 34 $\pm$ 3 years). The measurements of vibrotactile perception were performed over a wide frequency range (2–400 Hz) to permit separate assessment of the three different types of mechanoreceptors—Pacinian corpuscles, Meissner corpuscles, and Merkel disks. The gap perception was measured with a newly improved aesthesiometer. Preliminary analysis of the results indicates that measurements of vibrotactile perception particularly can detect sensory changes in the fingers not consistently found by conventional clinical tests.

WEDNESDAY AFTERNOON, 13 MAY 1987 — MT. RAINIER, 1:00 TO 2:30 P.M.

Session V. Noise IV: Noise Control Methods and Applications and Effects of Noise (Poster Session)

Angelo Campanella, Chairman
Campanella Associates, 3201 Ridgewood Drive, Columbus, Ohio 43220

All posters will be displayed from 1:00–2:30 p.m., and contributors will be at their posters the entire time.

Contributed Papers

V1. Noise control case history: Cafeteria environment. Larry H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910)

Complaints from employees at a local research institution about difficulties in communicating while eating lunch in the Institute's cafeteria led the institution to request recommendations as to possible facility modifications. Premodification sound surveys were conducted with and without the employees present. Recommendations were then made to management as to facility modifications that would be expected to significantly lower the cafeteria's sound level during the peak lunch period. After the recommendations were implemented, sound surveys were conducted to verify the predicted levels of noise reduction. The facility modifications included extensive use of sound-absorbing wall panels, a lowered and improved acoustical ceiling, and the construction of a partial barrier between the food-serving line and the eating area. At maximum usage, a 12 dB(A) (A-weighted sound-pressure level) reduction in the cafeteria's sound level was achieved [from 75 to 63 dB(A)]. No additional complaints have been received from the Institute's employees regarding noise in the cafeteria.

V2. Application of the STSF system to noise source identification in domestic appliances. U. D. Dietschi and J. S. Bolton (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

The acoustic energy produced by small appliances often originates at one point within the device (e.g., an electric motor) but actually radiates from a number of points on the casing. It is useful to identify these surface sources since they may indicate internal energy transmission paths; however, the task is made difficult by the fact that they are inevitably correlated and are often closely spaced. Recently the newly developed Brüel & Kjaer spatial transformation of sound fields (STSF) system was used to identify sources in this situation. This system is a broadband adaptation of the acoustical holography technique; it makes use of sound-pressure measurements at a number of reference locations and at an array of field points to identify the wavenumber components of the field. These may be used to reconstruct the entire sound field exterior to a plane tangent with the source. By examining intensity maps on orthogonal planes near the source, closely spaced individual surface sources may be identified. The application of this technique to a complete noise control procedure will be illustrated in this paper.

V3. Determination of vehicle noise source levels. Richard C. Smart and Stanley E. Dunn (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

In 1974 the Florida Department of Transportation participated in a study sponsored by the Federal Highway Administration to determine vehicle source levels. This paper reports the findings of a second study designed to extend the range of the operating speeds of the vehicles in question down to 20 mph (32 kph). A total of 11 sites were included in the study with vehicles operating between 20 to 61 mph. The subsequent data were analyzed to provide distributions of maximum passby sound levels and reference energy mean emission levels as a function of speed for automobiles, medium trucks, and heavy trucks. The data were analyzed over the entire speed range and on the basis of 5 mph speed band to provide empirical relations for use in predictive models. Of particular interest was the extent to which the choice of these formats influenced the noise source relationships and, of more significance, the extent to which the introduction of low-speed data which were not previously available influenced source level relationships. [Work supported by the Florida Department of Transportation.]

V4. A computer-based noise measurement system. Patrick Luquet (Sœur-Anne Corporation, 43, Avenue de la République, Villeurbanne, France, 69100)

This paper describes a computer-based measurement system that enables the operator to identify different noise sources and to measure their separate acoustical contribution to noise buildup. The principle is based upon the storage of 131 072 short-term L_{eq}'s (through a sound level meter with built-in memory) that are processed by personal computer. The method is illustrated by case studies ranging from urban noise surveys to industrial noise exposure monitoring, and includes environmental noise mapping. It is shown that both the method and the instrument have made it possible to reduce study costs while increasing the number and duration of sampling measurements, improving reliability factor of analyzed results, and allowing for the identification and quantification of multiple noise sources. The data-processing software presented in this paper provides the means to speed up report production while guaranteeing minimum-error data interpretation.

V5. A method for presenting dynamic performance characteristics of highly damped vibration isolators. Neil G. Plesner (E-A-R Division/Cabot Corporation, 7911 Zionsville Road, Indianapolis, IN 46268-0898)

A method for presenting dynamic performance characteristics of highly damped vibration isolators is discussed. The method utilizes normalized stiffness versus load graphs obtained from laboratory test data and enables determination of dynamic stiffness and natural frequency as a function of isolator load and temperature. Isolators examined were grommets and bushings made from three PVC thermoplastics. Because similar types of isolators show similar isolation characteristics with respect to load and temperature, they can be categorized into families based upon material formulation and isolator geometry. This provides a means for presenting a large amount of performance information in an abbreviated format. Methods of data acquisition and reduction are discussed briefly with major emphasis on the determination of isolator effectiveness for basic noise and vibration control problems.

V6. Ambulance sirens and in-vehicle noise exposure: An investigation into alternative siren speaker placement. Martin W. Van Buren (Ambulance Service, ADABEC 825, Hennepin County Medical Center, 701 Park Avenue, Minneapolis, MN 55415) and Gary Caple (Federal Cartridge, Anoka, MN 55303)

Two modular Road Rescue type III ambulances on 1985 Ford E350 XL RV cutaway vans, cab headliners insulated with fiberglass, modules with $1\frac{1}{2}$ in. of self-extinguishing polyurethane foam planking, were, respectively, outfitted with Southern D-50-C chrome bell 100-W speakers, forward facing, light-bar mounted above cab at 15° off center, speakers approximately 2.3 m from ground (condition 1) and with Whelen SA-450G speaker strobe 100-W speakers, forward facing, flush-radiator-grill-mounted, 0.94 m apart, speakers approximately 0.9 m from ground (condition 2). Each vehicle was placed on tarmac with sirens set on "wail" mode and SPLs were taken 1.2 m from ground in front of the vehicles along a 180° arc at points up to 30.5 m and in the vehicle at ear level (referenced to driver, passenger, patient, etc.). SPLs measured outside the vehicle at 30.5 m showed the grill-mounted speakers typically produced 14 dBA more intensity than the conventionally light-bar-mounted speakers at 0° azimuth. SPLs inside the vehicle were 11 to 16 dBA quieter in condition 2 depending on position measured. Efficacy of siren itself for emergency alerting was enhanced and occupant noise exposure was reduced when sirens were grill mounted.

V7. Effect of noise in ICU on heart rate and annoyance in adult postoperative patients. Carol F. Baker (College of Nursing, The Ohio State University, Columbus, OH 43210)

The relationships of environmental noise in a surgical ICU, measured as sound pressure levels in dBA, to changes in heart rate and perceived annoyance were determined in 28 postoperative adult patients. Levels were recorded for a continuous 6-h period on calibrated graph paper. Heart rate was recorded simultaneously by converting ECG signals from the bedside monitor. Data were sampled every 12 s and a mean obtained within each noise episode. The graph patterns were classified into four types of noise: talking inside the room, talking outside, nontalking, and ambient. The means within each type were averaged. Annoyance to ICU noise, which included disturbance from sound, disturbance to activities and to sleep, was measured by a questionnaire. Intervening variables of noise sensitivity and hearing acuity were also determined for each subject. Results showed noise levels in patients' rooms ranged from 60–64 dBA. A high ambient level was created by the use of oxygen which masked outside sounds. Sixty-eight percent of subjects showed a significant ($p = 0.10$) increase in heart rate with an increase in decibel from a previous noise level. Mean heart rate was greatest during talking inside the room. Heart rate was second highest during nontalking noise. Heart rate during ambient conditions was highly variable within subjects. The majority of subjects reported little annoyance to ICU noise, except when it interfered with sleep. Noise sensitivity and mild hearing loss were not significant factors in annoyance to ICU noise.

V8. Comparison of hearing acuity of rural, urban, and industrial populations. S. Raja and K. Harjeet Singh (Ergonomics Division, National Institute for Training in Industrial Engineering, Vihar Lake Road, Bombay-400 087, India)

Auditory acuity of 104 rural individuals, 126 urban individuals not exposed to occupational noise, and 118 workers from a textile mill was examined in order to study the influence of noise exposure characteristics on hearing level of the three populations. Pure-tone audiometry in 5-dB steps at selected frequencies was conducted with the help of a portable diagnostic audiometer. Noise level recordings were made by a B&K impulse precision sound level meter and an octave band analyzer. The differences in hearing level at 4 kHz between the populations were found to be statistically significant, with the rural individuals exhibiting superior hearing capacity compared to other groups, and urban individuals showing better ability compared to their industrial counterparts. In the low frequencies (0.5 and 1 kHz), however, significant differences were observed only between rural and industrial workers. The comparison between urban and industrial workers yielded only differences in absolute values which were not significant. Some factors contributing to the poorer hearing capacity of urban individuals have been discussed. The significance of this evaluation in the context of a mass movement of individuals from rural to urban areas to seek employment has been outlined.

Session W. Underwater Acoustics IV: Seabed Acoustics

Ian Roebuck, Chairman
Admiralty Research Establishment, Portland, Dorset DTS-2JS, United Kingdom

Chairman's Introduction—1:00

Contributed Papers

1:05

W1. Measurement of sound absorption and sound speed in terrigeneous sediments. Marco M. P. Weydert, Nicholas Murray, and Marco D'Alessandro (Commission of the European Communities, Joint Research Centre—Ispra Establishment, 21020 Ispra (VA), Italy)

The absorption and speed of sound were measured between two bore holes each 150 m deep and spaced 4 m apart. Sediment profile samples have been obtained from the two wells and the water content, porosity, density, and mineralogy determined. Acoustic measurements were made at selected depths in the water saturated sediments down to a maximum of -83 m. The acoustic frequencies ranged from 7.5–18 kHz; at the lower frequency, the absorption varied from 0.3 dB/m in clay to 0.4 dB/m in sand. At 18 kHz the corresponding values were 2.0 and 4.5 dB/m, respectively. Measurements of sound speeds for clay resulted in a velocity of $1480 + 15$ m/s, 1540–1820 + 15 m/s in silt, and $1540 + 15$ m/s for sand, at a temperature of 12 °C. The absorption measurements compare well at 12 kHz to those made in late 1986 in deep ocean sediments using penetrators that were able to penetrate up to 55 m into the sediments.

1:20

W2. Deep ocean/seabed–satellite relay for long-term quasi-real-time data transmissions. C. N. Murray and M. M. P. Weydert (Commission of the European Communities, Joint Research Centre—Ispra Establishment, 21020 Ispra (VA), Italy)

During the last 2 years, work has been underway at the Joint Research Centre of the Commission of the European Communities on the use of satellite communications for the transmission of data from instrumentation placed in the ocean bed. The setup of the transmission link has required the development of several new systems. These include an underwater vehicle capable of successfully emplacing instrument packages within deep ocean sediment formations, a transmitter–receiver system whose characteristics are such as to be able to send acoustic signals through both 50 m of sediments and a 6-km water column, sensors which will give information on sediment characteristics, and a satellite communications system for the automatic quasi-real-time relay of data from the emplaced instrumentation to the laboratory in Ispra (Europe). The present paper will discuss the acoustic link and the relay/buoy system and its applications to oceanography and meteorology.

1:35

W3. Low-frequency acoustic/seismic propagation in a sloping ocean environment: Comparison between measured results and numerical predictions. Hassan B. Ali and Michael F. Werby (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004)

The propagation of low-frequency signals and noise in shallow water, and more generally in a bottom-limited environment, is of considerable interest to the Navy. A particular aspect of this problem, namely the relative merits of waterborne versus seismic propagation paths, has been investigated using the results of at-sea measurements and predictions based on geoacoustic and numerical models of the test environment. The experiment was conducted by the Naval Ocean Research and Development Activity in a shallow water region off the southeastern coast of the U. S. A vertical array of hydrophones in the water column and a distribution of tri-axial geophones on the ocean bottom were used to monitor the response to both cw and broadband (explosives) sound sources. Predictions of the propagation were made using the SAFARI version of FFP, the wide-angle version of the Parabolic Equation Method, and the SNAP normal mode model. Preliminary results of the measurements are discussed and comparisons are made with the predictions based on the numerical models.

1:50

W4. Laboratory measurements of geo-acoustic properties of the New Jersey Shelf sediments. Mohsen Badiey, Tokuo Yamamoto, and Altan Turgut (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

Laboratory measurements of selected geo-acoustic properties of sediment samples collected at the New Jersey Shelf are presented. A high precision torsional resonant column apparatus was used to determine both the shear modulus and the attenuation at different confining stress conditions. Porosity and permeability of these sediments were also measured. These measurements are compared to both the existing geological data in this region and the remote measurements of the shear modulus using the Bottom Shear Modulus Profiler (BSMP). Finally, the quality factor and the attenuation are calculated using measured properties as input to a numerical model for acoustic wave propagation in shallow water. [Work supported by ONR.]

2:05

W5. Horizontally polarized shear waves in shallow marine sediments. George H. Sutton (Rondout Associates, Inc., P. O. Box 224, Stone Ridge, NY 12484), John I. Ewing (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), Noël Barstow, and Jerry A. Carter (Rondout Associates, Inc., P. O. Box 224, Stone Ridge, NY 12484)

In June 1986 a seismic experiment emphasizing the observation of shear waves in the bottom sediments was conducted off the southern New Jersey coast in water depths between about 10 and 56 m. This paper emphasizes the generation and detection of horizontally polarized shear waves *SH* produced by a pair of air (mud) guns, mounted to apply horizontal forces, on a sled deployed from the ship and capable of being dragged along the bottom. Signals were digitally recorded from three-component geophones and a hydrophone. Shear waves and boundary waves were recorded in the frequency band 2 to 40–50 Hz having group velocities ranging from about 50 to 300 m/s. Both *SH*/Love wave and *P*/*SV*/Rayleigh–Stoneley–Scholte wave signals were clearly recorded and resolved by combining sums and differences of two properly oriented

WEDNESDAY PM

horizontal components and comparing the resolved horizontal data with vertical and pressure components. Full-waveform synthetic seismograms are being used to aid inversion for shear velocity and attenuation models that are required to be consistent with information available from test borings as well as the shear wave data. [Work supported by ONR.]

2:20

W6. The acoustics of "black smoker" hydrothermal plumes. David R. Palmer and Peter A. Rona (NOAA/AOML, 4301 Rickenbacker Causeway, Miami, FL 33149)

High-temperature "black smoker" hydrothermal plumes occur when seawater that has penetrated into the oceanic crust and assimilated heat from magma is discharged from vents located at the axis of a mid-ocean ridge. The acidic, metal-rich, discharge mixes with alkaline, oxidizing seawater, and a fine suspension of sulfide particles is precipitated and convected by the flow. Vents fields have now been found at both fast and slow seafloor spreading centers and may be a ubiquitous feature of mid-ocean ridges. A review of the progress made in using underwater acoustics to study black smoker plumes is presented. Both active and passive techniques are being investigated. Active techniques involve a high-frequency monostatic sonar mounted on a submersible. Analysis of the amplitude and phase of the signal backscattered from the plume provides information about the three-dimensional shape of the plume as well as estimates of the flow-velocity field of the discharging fluid. Passive techniques use bottom-mounted hydrophones to listen to the very low-frequency, hydrodynamic noise generated by a plume. These noise signatures have potential use in locating, characterizing, and monitoring plume sites and in determining the contribution plume noise makes to the overall ambient noise field in the ocean.

2:35

W7. Exact reconstruction of ocean bottom velocity profiles from monochromatic scattering data. Andre A. Merab (MIT/WHOI Joint Program in Oceanography/Oceanographic Engineering, Woods Hole, MA 02543) and George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A technique for exactly reconstructing ocean bottom velocity profiles from monochromatic scattering data is presented. The method involves an adaptation of the Gelfand–Levitan inversion procedure, originally developed in the context of quantum mechanical potential scattering, to the problem of determining the acoustic properties of the bottom. In this approach, if the bottom plane-wave reflection coefficient at a fixed frequency is known for all angles of incidence, then the sound velocity versus depth can be uniquely recovered. If the reflection coefficient at all angles is known at two frequencies, then both the density and velocity profiles can be uniquely reconstructed. The inversion scheme consists of Fourier transforming the reflection coefficient and solving the Gelfand–Levitan integral equation. This procedure is implemented numerically and applied successfully to synthetic data for several typical profiles. The stability of the method in the presence of noise is also demonstrated. [Work supported by ONR.]

2:50

W8. Determination of modal attenuation coefficients and the attenuation profile of the bottom in shallow water. Subramaniam D. Rajan, James F. Lynch, and George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

Techniques for determining modal attenuation coefficients and the compressional wave attenuation profile of the bottom in shallow water are presented. The required input data consist of measurements of the pressure field versus range due to a monochromatic point source. Several methods are described for obtaining modal attenuation coefficients from the pressure field or its Hankel transform (the depth-dependent Green's function versus horizontal wavenumber). These modal attenuation coefficients can be related to the bottom attenuation profile through an integral equation that can be solved using linear inverse theory. The method is demonstrated using synthetic noise-free and noisy data. The results of inverting experimental data from Nantucket Sound at 140 and 220 Hz are also described. Resolution and variance estimates for the bottom attenuation profile are presented. [Work supported by ONR.]

WEDNESDAY AFTERNOON, 13 MAY 1987 — CANYON HALL, 1:15 P.M.

Meeting of Accredited Standards Committee S2 on Mechanical Shock and Vibration

to be held jointly with the

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

J. C. Barton, Chairman S2
Caterpillar Tractor Company, Research Department, 100 N. E. Adams, Peoria, Illinois 61629

G. Booth, Chairman, Technical Advisory Group for ISO/TC 108
220 Clark Avenue, Brandford, Connecticut 06405

Standards Committee S2 on Mechanical Shock and Vibration. Working group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory Group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those committees) including a report on the meeting of ISO/TC 108, held in Washington, DC from 30 March to 10 April 1987.

Session X. Physical Acoustics IV: Special Session on Sonic Energy for Particle Processing

Osman K. Mawardi, Chairman
Electrical Engineering and Applied Physics Department, Case Western Reserve University, Cleveland, Ohio 44106

Invited Papers

1:15

X1. Solid particles in sonic field. Osman K. Mawardi (Electrical Engineering and Applied Physics Department, Case Western Reserve University, Cleveland, OH 44106)

A review is presented for the dynamics of solid particles in suspension in a fluid and subjected to sonic energy. The controlling parameters associated with collective behavior of the particles such as precipitation or coagulation are identified. A brief discussion is then given on a classification of observed phenomena in the processing industry and by means of these parameters.

1:35

X2. Aerosol agglomeration in high-intensity acoustic fields. David T. Shaw and Sushil Patel (Department of Electrical and Computer Engineering, State University of New York at Buffalo, Amherst, NY 14260)

The operational conditions for the application of acoustic agglomeration (AA) as a preconditioner for the control of particulate matter under high temperature and pressure are discussed based on recent data on the acoustically induced turbulence. The AA kernels in various regimes are evaluated for various performance parameters, including acoustic intensity, frequency, residence time, gas temperature, and pressure. The turbulent interaction is found to be the dominant process for particle agglomeration. Since AA is not particle-size dependent, it is ideally suited for preconditioning, and subsequent removal of the submicron particles. A computer model for the prediction of AA has been developed and good agreements have been found between the theory and experimental data. Based on a comprehensive evaluation of the techniques for the generation of high-intensity sonic field and problems in their applications, a dirty-air chopper has been built and tested with very favorable results.

2:05

X3. Acoustic agglomeration of power plant fly ash for environmental and hot gas cleanup. Gerhard Reethof (Department of Mechanical Engineering, The Pennsylvania State University, University Park, PA 16802)

Emission of small particulates from coal-fired power plants in the range of 0.5 to 5 μ results in particularly serious injury to the human respiratory system. Currently used environmental cleanup devices in power plants such as bag houses, electrostatic precipitators, wet scrubbers, and others are very efficient in removing particulates which are larger than about 2 μ in size, leaving by far the majority of these tiny particulates to be emitted into the air environment. Current regulations for effluent cleanup emphasize the mass removal without any attention to the particulate size. Acoustic agglomeration is a means of causing these tiny particulates to collide with one another, adhere, and thereby result in the formation of larger particles, thus removing the tiny particulates from the aerosol stream. In this manner, the efficiency of particle removal of the conventional removal devices can be substantially enhanced. This paper introduces some of the fundamental aspects of acoustic agglomeration theory and practice and presents some of the results of recent studies.

2:35

X4. Removal of particulate contaminants from surfaces. K. W. Montz (Department of Chemical and Materials Engineering, 1153 EB, The University of Iowa, Iowa City, IA 52242), P. B. Butler (Department of Mechanical Engineering, The University of Iowa, Iowa City, IA 52242), and J. K. Beddow (Department of Chemical and Materials Engineering, 1153 EB, The University of Iowa, Iowa City, IA 52242)

An experimental investigation was made on the interaction of an acoustic field with particulate matter adhered to a substrate. The particles of interest are in the μ-size range. In the experimental design, a combined acoustic/convective field is used to overcome the adhesive forces of the contaminant materials. A key parameter in this work is the removal efficiency (based on number counts) of contaminant particles from the surface of materials. An image analysis technique is used to determine particle counts before and after the process. The relative adhesive and removal forces are a function of particulate material and substrate properties. Preliminary results using a factorial design illustrate the influence of several acoustic properties on the removal efficiencies.

Contributed Paper

3:00

X5. Acoustic suspension processing. D. L. Feke (Chemical Engineering Department, Case Western Reserve University, Cleveland, OH 44106) and D. Hazony (Electrical Engineering and Applied Physics Departments, Case Western Reserve University, Cleveland, OH 44106)

Separation and purification of fine solids suspended in fluids are involved in a variety of technologies. A process utilizing high-energy acoustic planes that are separated by low-energy zones will be described for these purposes. Results of theoretical and experimental studies will be discussed.

WEDNESDAY AFTERNOON, 13 MAY 1987 CELEBRATION HALL, 1:30 TO 2:35 P.M.

Session Y. Musical Acoustics II, Architectural Acoustics IV, and Shock and Vibration II: Distinguished Lecture on Early Lateral Reflections in Modern Concert Halls

Thomas D. Rossing, Chairman

Department of Physics, Northern Illinois University, Dekalb, Illinois 60115

Since Professor Cremer will be unable to be present, his paper will be read for him by Theodore Schultz, Theodore J. Schultz Associates, Boston, MA 02118.

Chairman's Introduction—1:30

Invited Paper

1:35

Y1. Early lateral reflections in modern concert halls. Lothar Cremer (D-816 Miesbach, Federal Republic of Germany)

It is agreed that part of the acoustical qualities of the "shoe box" halls of the 19th century results from their nearly parallel sidewalls and from "cue ball" reflections below the soffits of their balconies. This arrangement conflicts with the enlargement of the stage and audience capacity and with the modern expectation of comfortable seating and full view of the stage from each location. Three possibilities for overcoming these difficulties, based on the author's recent consulting work, are discussed: (a) Placement of a large part of the audience behind the stage but not lateral to it (Madrid, under construction). This solution is unsatisfactory for visual performances; (b) providing strong early reflections from proscenium towers which focus the sound in the vertical plane but diffuse it in the horizontal. This solution, proved by echograms, can be used in even a very broad (26-m) stage, such as the Salle Apollon, Nice, opened in 1985. Subdivision of the audience by "vineyard" steps produces early lateral reflections; (c) the use of vineyard steps, allowing an optimal and simple "trapezium terrace" system (Las Palmas, Gran Canaria, planned). Here, the visually preferred diverging fan-shaped plan is compensated acoustically by the converging steps of the trapezium terraces, which could easily be inclined so that they provide early lateral reflections to the middle of the terraces. The limits of this geometrical behavior are quantitatively investigated according to Fresnel's formula. The more diffuse reflections at lower frequencies are profitable for the late reverberant sound.

Session Z. Psychological and Physiological Acoustics VI: Masking

Robert H. Gilkey, Chairman
Central Institute for the Deaf, 818 South Euclid, St. Louis, Missouri 63110

Contributed Papers

1:30

Z1. A model of the effects of signal delay and masker level on masked measures of frequency selectivity. Robert A. Lutfi (Waisman Center, University of Wisconsin, Madison, WI 53705)

In a previous paper [R. A. Lutfi, J. Acoust. Soc. Am. **76**, 1045–1050 (1984)], the following relation was used to predict measures of frequency selectivity obtained in forward masking from measures obtained in simultaneous masking: $F(g) = G + H(g) - H(0)$, where, for a given masker level, F is the amount of forward masking (in dB) as a function of signal-masker frequency separation (g), H is the amount of simultaneous masking, and G is the amount of forward masking for $g = 0$. In the present study, the relation was tested for a wider range of signal and masker frequencies, masker levels, and signal delays. The relation described thresholds from all conditions well with the inclusion of one free parameter λ corresponding to a constant frequency increment: $F(g) = G + H(g+\lambda) - H(\lambda)$. The parameter λ was required to account for observed shifts in the frequency of maximum forward masking. It is suggested that a single tuning mechanism can account for commonly observed differences between simultaneous- and forward-masked measures of frequency selectivity. [Work supported by National Science Foundation Grant BNS 83-08498.]

1:45

Z2. Roving level tone in noise masking. Gerald Kidd, Jr., Christine R. Mason, Merry A. Brantley, and Grace A. Owen (Department of Communication Disorders, Boston University, 48 Cummington Street, Boston, MA 02215)

The results of tone in noise detection experiments have long formed an empirical basis for the notion of the "critical band" and have led to the development of the "energy detection" model. In this study, the detectability of tones in bands of random noise was measured for conditions in which the overall level of the noise was roved randomly from interval to interval of each experimental trial. The functions relating masked tone threshold to noise bandwidth for the roved conditions were identical to those obtained when no rove was employed at "supracritical" bandwidths. At "subcritical" bandwidths, the detection thresholds were higher for the roved conditions, but were still lower than would be predicted from arguments based purely on energy detection. These results suggest that the traditional critical band and energy detection models should be modified to account for discrimination based on spectral shape or waveshape. [Work supported by NIH and NMRDC.]

2:00

Z3. Effect of masker bandwidth on the detection of signals at unexpected frequencies. Huanping Dai, Bertram Scharf, and Søren Buus (Auditory Perception Laboratory, Northeastern University, Boston, MA 02115)

In wideband noise, listeners tested with the probe-signal method [G. Z. Greenberg and W. D. Larkin, J. Acoust. Soc. Am. **44**, 1513–1523 (1968)] detect a signal at an expected frequency (primary) while missing signals at a far, unexpected frequency (probe). The present experiments test the hypothesis that the internal filter is narrowest at the primary frequency, thereby admitting less external noise there. According to this hypothesis, the probe should be as detectable as the primary when the external masker is narrower than the internal filter. Against wideband noise, subjects detected a 1-kHz primary on 90% of the trials and a 2-kHz probe on 60% of the trials. Against two simultaneous subcritical noises—one at 1 kHz, the other at 2 kHz—results were similar. With the noise at 2 kHz narrowed to 50 Hz, 6 of 12 subjects detected the probe reliably, possibly because the strong pitch of the masker drew attention to the 2-kHz probe. Overall, these results suggest that attention does not affect the width of the internal filter. [Work supported by NIH.]

2:15

Z4. Psychometric functions for multicomponent maskers with spectral uncertainty. Donna L. Neff and Brian P. Callaghan (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131)

Previous experiments used a two-alternative, forced-choice, adaptive procedure to measure thresholds for a 1000-Hz signal in the presence of maskers composed of multiple sinusoids. These sinusoids were drawn at random from a 300- to 3000-Hz range, excluding the signal frequency and frequencies within a critical band around the signal. This experiment examined psychometric functions for 50 of the original 200 maskers with 10 components, using signal levels from 10–80 dB SPL. The psychometric functions were well fitted in d' by signal level coordinates, but individual functions typically spanned a 30- to 40-dB range. Corresponding adaptive thresholds did not differ from thresholds based on 200 maskers and were in agreement with the average of the predicted thresholds across the 50 maskers. Although individual maskers could differ in effectiveness by 10–15 dB, the majority produced similar masking. Overall, the adaptive procedure adequately characterizes performance in these conditions. [Work supported by NIH.]

2:30

Z5. Exact jnd functions for a generalized McGill–Goldberg counting model. William S. Hellman (Department of Physics, Boston University, Boston, MA 02215) and Rhona P. Hellman (Department of Psychology, Northeastern University, Boston, MA 02115)

In an earlier work [W. S. Hellman and R. P. Hellman, J. Acoust. Soc. Am. Suppl. 1 **79**, S34 (1986)], a generalization of the McGill–Goldberg counting model was presented. It was demonstrated that loudness functions, consistent with experiments, could be determined from intensity jnd functions that obeyed both the power-function near miss and Weber's law. The central formula for the model is the integral relation $N(I)^{1/2} = (h/2)\int dI/[IJ(I)] + a$, where h and a are constants, and $N(I)$ and $J(I)$ are the neural count and input jnd functions, respectively. Following McGill and Goldberg, the model was developed using a first-order approximation for the jnd induced change ΔN. To determine just how appropriate the first-order approximation might be, the results of bootstrapping the model are shown. That is, given the derived neural-count functions, the "exact" associated jnd functions are generated. For a power-function near miss as original input, an "exact" jnd function that exhibits a low-intensity deviation from power-function behavior is obtained. When the input jnd function obeys Weber's law, it is found that the bootstrapped jnd function is constant over a wide range of intensities. These results are compatible with psychophysical data. [Work partially supported by the Rehabilitation Research and Development Service of the VA.]

2:45

Z6. Comodulation masking release (CMR) as a function of signal frequency, flanking-band frequency, masker bandwidth, and flanking-band level. Gregory P. Schooneveldt and Brian C. J. Moore (Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England)

Thresholds for 400-ms signals were measured in the presence of a continuous narrow-band noise centered at signal frequencies (fs) ranging from 250–4000 Hz in 1-oct steps. The masker was presented either alone or together with a second band of noise (the flanking band) whose envelope was either correlated with that of the on-frequency band or was uncorrelated; its frequency ranged from 0.5 to 1.5 fs. CMR was defined as the difference between thresholds for the correlated and uncorrelated conditions. The CMR showed two components: a broadly tuned component occurring at all signal frequencies and all flanking-band frequencies, and a component restricted to flanking-band frequencies close to fs, which increased in magnitude with increasing fs. The second component was probably not a true CMR, but resulted from "beating" between the carrier frequencies of the two masker bands. Additional experiments, in which the bandwidth of the masker and the level of the flanking band were varied, support this interpretation. The first component, which is probably a "true" CMR, was only about 3 dB. [Work supported by the MRC, U. K.]

3:00

Z7. CMR for complex signals. Joseph W. Hall, III, John H. Grose (Division of Otolaryngology, Department of Surgery, University of North Carolina, Chapel Hill, NC 27514), and Mark P. Haggard (Institute of Hearing Research, University of Nottingham, Nottingham NG7 2RD, England)

Previous CMR investigations have examined the effect of across-frequency masker envelope coherence on the detection of a pure-tone signal. The present study extends this investigation to examine the effect of the number of components comprising the signal. The masker was composed of either one, two, or three 30-Hz-wide narrow-band noise components. In multiband conditions, the bands had either correlated or uncorrelated envelopes. For the one-component masker, the signal was a pure tone at the center frequency of the masker. For the two-component masker, the signal was either a one-component or a two-component signal. For the three-component masker, the signal was either a one-, two,- or three-component signal. For the three-component masker, CMR was greatest for the single-component signal. However, CMR was appreciable for two- and three-component signals. CMR generally increased as a function of increasing number of masker bands. Results will be discussed in terms of models of CMR. [Work supported by AFOSR.]

WEDNESDAY AFTERNOON, 13 MAY 1987 REGENCY BALLROOM A & B, 1:30 TO 2:50 P.M.

Session AA. Speech Communication V: Communication Disorders (Poster Session)

Mary E. Beckman, Chairman

Department of Linguistics, Ohio State University, Columbus, Ohio 43210

All posters will be displayed from 1:30–2:50 p.m. To allow all contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30–2:10 p.m., and contributors of even-numbered papers will be at their posters from 2:10–2:50 p.m.

Contributed Papers

AA1. Development and testing of artificial low-frequency speech codes. C. M. Reed, M. H. Power, K.K. Foss, N. I. Durlach, and L. D. Braida (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Previous work on frequency lowered speech for listeners whose hearing is confined to low frequencies has focused on signal processing of natural speech. In the present research, the use of artificial low-frequency codes for such listeners is explored. This research will help separate limitations imposed by signal processing schemes from limitations imposed by the listener's perceptual system. It will also provide background for the design of new systems based on the use of speech-recognition devices as front ends. Artificial frequency-lowering codes were developed for 24 consonants (C) and 15 vowels (V) for two values of low-pass cutoff frequency F (300 and 500 Hz). Individual phonemes were coded by a unique acoustic signal confined to frequencies less than or equal to F. The ability of normal-hearing listeners to identify coded C's and V's in fixed-context syllables was compared to their performance on single-token sets of natural speech utterances low-pass filtered to equivalent values of F. Performance on coded C's was superior to that on low-pass filtered speech for both values of F (by roughly 20 percentage points). An advantage for coded V's (of roughly 15 percentage points) was observed only at $F = 300$ Hz. Identification of C and V components in roving-context CV syllables was generally inferior to that obtained in the fixed-context conditions. [Work supported by NIH.]

AA2. Spectral shaping options for persons with low- or high-frequency sensorineural hearing loss. Donald J. Schum and M. Jane Collins (Division of Communication Disorders, Louisiana State University, 163 M&DA, Baton Rouge, LA 70803)

When hearing loss is limited to one particular frequency region, amplification is traditionally provided in the region of the poorest audiometric thresholds. To the extent that audiometric data correlate with auditory system damage, the usefulness of information provided to these regions of greatest damage is questionable. Optionally, gain may be provided to frequency regions where thresholds are improving towards normal. To that end, continuous discourse intelligibility ratings and consonant discrimination scores were obtained from individuals with either low-frequency or precipitously sloping high-frequency sensorineural hearing loss. During testing, spectral emphasis was variously placed in

regions of hearing impairment and/or improving thresholds. For the low-frequency impaired subjects, consistently inferior subjective ratings and discrimination scores were obtained when spectral emphasis was placed solely in the region of hearing loss. For the high-frequency impaired subjects, no single spectral weighting scheme emerged as superior or inferior. The current results are consistent with previous findings which have suggested that pure-tone thresholds may underestimate the degree of apical region damage in low-frequency sensorineural hearing loss. [Work supported by DRF.]

AA3. A multichannel tactile aid reduces articulation errors during spontaneous and imitative speech. William J. Gavin, Melinda K. Mathay, and Debra Harr (Institute of Logopedics, 2400 Jardine Drive, Wichita, KS 67219)

Five tactually experienced, hearing-impaired children were tested in two sessions wearing the tactile aid during both sessions but activated during only one session. Subjects received auditory and visual information throughout. Each session consisted of a spontaneous and an imitative speech task, each eliciting six productions of four monosyllable and four bisyllabic words. Recorded responses were transcribed using broad IPA techniques. Four of the five children produced fewer errors with the aid active compared to the nonactive condition during both the spontaneous and imitative tasks. The mean percent of errors in nonactive and active conditions for spontaneous task was 27% and 26%; for imitative task, 19% and 17%. A 2×2 randomized block analysis-of-variance test showed a significant main effect for task ($p < 0.001$). The tactile aid condition approached but did not reach significance ($p < 0.10$). The data are extremely encouraging. While the act of producing speech is largely a habitual motor response, removal of the tactile information resulted in a measurable increase in articulation errors.

AA4. Vocabulary training via a tactile communication aid with a prelingual deaf child. Robert E. Woodard and William J. Gavin (Institute of Logopedics, 2400 Jardine Drive, Wichita, KS 67219)

A wearable, self-contained 24-channel vibrotactile vocoder was used to train receptive vocabulary. The subject was a 6-year-old, prelingual, profoundly hearing-impaired female of average intelligence. Receptive vocabulary training was conducted a minimum of $\frac{1}{2}$ h/day, 4 days/week, with the subject receiving auditory, visual (speechreading), and tactile stimulation during training. Assessments of vocabulary growth were conducted every fifth session via a picture pointing task. Words for training were selected from Benedict [H. E. Benedict, "Language comprehension in 10- to 16-month-old infants," unpublished doctoral dissertion, Yale University, New Haven, CT (1976)] and clinician suggestions. After approximately 34 h of training over 13 weeks, the subject learned 60 words, a learning rate of 1.79 words/h. This rate is slower than the rate of 2.7 words/h reported for normal-hearing subjects [P. L. Brooks and B. J. Frost, J. Acoust. Soc. Am. **74**, 34–39 (1983)] but faster than the rate of 0.53 words/h reported for deaf subjects [S. Englemann and R. Rosov, Except. Child **41**, 243–253 (1975)].

AA5. Assessment of a tactile communication aid with a post-lingual deaf adult. Robert Rosov, Robert E. Woodard (Institute of Logopedics, 2400 Jardine Drive, Wichita, KS 67219), Dennis R. Ingrisano (University of Northern Colorado, Greeley, CO 80639), and Darlene Murrell (Humble I.S.D., Humble, TX 77338)

A case study was utilized to assess receptive vocabulary training using a wearable, self-contained 24-channel vibrotactile vocoder. The subject was a 58-year-old, post-lingual, adventitiously deafened female of average intelligence and above-average skills in learning word meanings (+2 s.d. on vocabulary subtest of the WAIS). It was predicted that assessing rate of vocabulary acquisition with this type of subject would be a more rigorous test of using tactile stimulation as a means of sensory substitution for the deaf than could be achieved using a prelingually deafened subject. The vocabulary training protocol was modified from Brooks and Frost [P. L. Brooks and B. J. Frost, J. Acoust. Soc. Am. **74**, 34–39 (1983)]. Training words were presented using only tactile stimulation with feedback. Training was conducted a minimum of 1h/day, 5 days/week. After approximately 88 h of training over 18 weeks, the subject was able to identify 15 words, yielding a word acquisition rate of 0.17 words/h. This differs considerably from acquisition rates previously reported of 2.7 words/h for normal-hearing subjects [P. L. Brooks and B. J. Frost, J. Acoust. Soc. Am. **74**, 34–39 (1983)] and 0.53 words/h for hearing-impaired subjects [S. Englemann and R. Rosov, Except. Child. **41**, 243–253 (1975)].

AA6. Auditory/phonetic categories in a patient using a multichannel cochlear implant. M. F. Dorman (Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85287) and Geary McCandless (Department of Otolaryngology, University of Utah Medical School, Salt Lake City, UT 84114)

Identification functions in response to stimuli from several acoustic continua from a patient using a Symbion multichannel cochlear implant have been obtained. Auditory/phonetic categories of normal configuration were obtained for stimuli which varied in event duration ("slit"–"split" and "chop"–"shop"), in steady-state frequency composition (/ɪ/–/ɛ/ and /s/–/ʃ/, and in event duration and low-frequency composition (/ga/–/ka/). For the /ga/–/ka/ contrast, $F1$ onset frequency and VOT "trade" to determine categorization. Abnormal categories were obtained for stimuli which varied in $F2/F3$ onset frequency (/ba/–/da/) and for stimuli which varied in signal amplitude (/s/–/θ/). These data indicate that, in at least some instances, the implant can provide auditory information with sufficient resolution to afford normal phonetic categorization.

WEDNESDAY PM

AA7. Using cued speech to clarify speechreading. Kathy L. Tonry (University of Cincinnati, Communication Disorders, ML 379, Cincinnati, OH 45221-0379)

Speechreading for hearing-impaired individuals is a monumental task due to visual similarities of some speech sounds and the lack of visibility of others. Cued speech is a visual supplement to speechreading used to remediate this problem. It employs a system of 12 visual hand cues. Four hand positions are used to clarify vowels and eight hand configurations are used to differentiate consonants which are visually unclear. Two previous studies by Ling and Clarke [D. Ling and B. R. Clarke, Am. Ann. Deaf **120**, 480–488 (1975); and B. R. Clarke and D. Ling, Volta Rev. **78**, 23–34 (1976)] evaluated the speechreading capabilities of hearing-impaired subjects using cued speech. Results from the first study (subjects used cues for 1 year) supported the use of cues for identifying words in phrases. The follow-up study (1 year later) supported the idea that cued speech facilitates speechreading for sentences, phrases, and words. The present informal study indicates improved language skills in nine hearing-impaired children using cued speech.

AA8. Performance on SPIN as a function of subject age. Melanie L. Matthies and Robert C. Bilger (Department of Speech and Hearing Science, 901 South Sixth Street, University of Illinois, Champaign, IL 61820)

The effect of aging on word-recognition ability typically has been confounded with the effect of hearing loss in studies involving the analysis of age-group means. A correlational approach allows for an efficient analysis of the increasing heterogeneity with age in word-recognition performance and the effects of hearing impairment. Correlational analyses of performance on SPIN and subject age are described for four experiments. The data sets include subjects from the SPIN standardization study (19–69 years), a hearing-aid fitting experiment (29–74 years), a study of older listeners (60–81 years), and subjects from a study of word recognition in noise (18–70 years). All subjects were tested at a single presentation level calculation +50 dB *re*: babble threshold but at a variety of signal-to-babble ratios (+ 8 to − 2 dB S/B). Correlations between SPIN scores and age were only occasionally significantly different from zero and were found to increase slightly with the signal-to-babble ratio used and to decrease with the severity of hearing impairment. [Work supported by NINCDS.]

AA9. Effects of contextual prosodic patterns on the auditory comprehension of normally stressed targets. Mikael D. Z. Kimelman (Department of Communication Disorders, L.S.U. Medical Center, 1900 Gravier Street, New Orleans, LA 70112) and Malcolm R. McNeil (Department of Communicative Disorders, University of Wisconsin—Madison, Madison, WI 53706)

Reports indicate that neurologically normal subjects produce faster reaction times to phoneme targets when the context preceding the targets is produced with a stressed versus normally stressed prosodic pattern. The effects of contextual prosodic patterns on the auditory comprehension of normally stressed targets by aphasic listeners were investigated. Target words were computer edited out of paragraph-length stimuli that had been produced with emphatic stress and with normal stress. The target words were then replaced with normally stressed cognates. Analysis revealed that aphasic auditory comprehension was better when the context surrounding the target words had been produced with an emphatically stressed rather than a normally stressed prosodic pattern. Acoustic analyses of the emphatically stressed and normally stressed stimuli revealed differences in duration but not peak fundamental frequency of words preceding target words. [Work supported by NICHD Core Grant #5 P30 HD03352 to Waisman Center, University of Wisconsin—Madison.]

AA10. Anticipatory coarticulation in aphasia: Acoustic and perceptual evidence. William F. Katz (Department of Psychiatry, M-031P, University of California—San Diego, La Jolla, CA 92093)

Acoustic analyses of anticipatory coarticulation were conducted for the initial consonants of CV [si su ti tu ki ku] and CCV [sti stu ski sku] productions by six normal and ten aphasic (five anterior, five posterior) subjects. For normal subjects' productions, reliable coarticulatory shift was found for almost all measurements, indicating that acoustic correlates for anticipatory coarticulation obtain for [s], [t], and [k] in a prevocalic environment as well as when [s] is the initial consonant of a CCV syllable. The data for the aphasic subjects were statistically indistinguishable from those of the normal group, with no differences noted as a function of aphasia type. In the perceptual experiment, a subset of the consonantal stimuli produced by the normal and aphasic subjects was presented to a group of ten naive listeners for a vowel identification task. Listeners were able to identify the productions of all subjects at a level well above chance. In addition, small but statistically significant group differences were observed, with the [sV], [skV], and [tV] productions by anterior aphasics showing significantly lower perceptual scores than those of normal subjects. The results suggest a model of speech programming in which coarticulatory processes are largely preserved in aphasia. [Work supported in part by grant NS-22282 to Brown University Department of Linguistics.]

AA11. Recent progress in computer recognition of cerebral palsy speech. J. R. Deller, Jr. (Electrical and Computer Engineering Department, Northeastern University, Boston, MA 02115), C. G. Venkatesh (Mathematics Department, Northeastern University, Boston, MA 02115), D. Hsu (Electrical and Computer Engineering Department, Northeastern University, Boston, MA 02115), L. J. Ferrier (Speech and Language Pathology and Audiology Department, Northeastern University, Boston, MA 02115), and M. B. Cozzens (Mathematics Department, Northeastern University, Boston, MA 02115)

This paper is ultimately concerned with the development of an AI communication aid for the nonverbal, profoundly motor disabled. Part of this effort is aimed at automated recognition of speech of the nonverbal. The speech study has focused on the cerebral palsied (CP) population, and recent papers have reported methodology and early experimental results concerning the lower (acoustic and word) level recognition tasks for CP speakers [Deller *et al.*, *Proceedings of the ASSP Workshop on Acoustics*, New Paltz, NY (1986) and Deller *et al.*, *Proceedings of ICASSP '87*, Dallas, TX (in press)]. The purpose of this paper is twofold: (1) to present further experimental data concerning the word level recognition problem, particularly with regard to vowels and diphthongs in varying consonant contexts, and (2) to present a graph-theoretic approach to employing sentence level grammar. The graph search strategy, based on the Planar separator theorem [R. J. Lipton and R. E. Tarjan, SIAM J. Appl. Math. **36**, 177–189 (1979) and Venkatesh *et al.*, *Proceedings of the IEEE International Conference on Circuits and Systems'87*, Philadelphia (in press)], is particularly useful in this problem where isolated regions of reliability in the message may be surrounded by unrecognizable acoustics. [Work supported by the W. R. Hearst Foundation, The United Cerebral Palsy Research and Education Foundation, and by N.I.H. under Grant No. 1R03RR02630.]

AA12. Tongue–jaw displacement variability in the fluent speech of stutterers and control subjects. P. J. Alfonso (University of Connecticut, Storrs, CT 06268 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510), B. C. Watson (University of Texas at Dallas/Callier Center, Dallas, TX 75200), and R. Story (University of Connecticut, Storrs, CT 06268 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

The x-ray microbeam installation at the University of Tokyo was used to compare supralaryngeal kinematic patterns of fluent utterances produced by a control subject and two stutterers. For the control subject, trial-to-trial variability in individual and combined tongue–jaw peak displacement for alveolar obstruent closure and release was equivalent to the variability in lip and jaw displacement for bilabial stop closure and release by a larger group of normal subjects [V. L. Gracco and J. H. Abbs, Exp. Brain Res. (in press)]. The stutterers demonstrated greater variability than the control in individual tongue and jaw displacement, and the combined tongue–jaw displacement variability was not less than the individual articulator variability. In contrast to the control, the stutterers achieved closure and release primarily by jaw displacement with little, and occasionally paradoxical, tongue displacement. The lack of covariability in tongue–jaw displacement, coupled with the dominance of a single member of an articulator complex, suggests that stutterers lack the same degree of precision and flexibility observed in normal speech motor systems to efficiently meet invariant object-level goals. [Work supported by NIH NS-13617.]

Plenary Session

Ira Dyer, Chairman
President, Acoustical Society of America

Business Meeting

Presentation of Awards

Gold Medal to Cyril M. Harris
R. Bruce Lindsay Award to Ilene J. Busch-Vishniac

Musical Entertainment

The Cardinal Quartet, a string quartet of precocious students from the Indiana University School of Music, will perform three pieces by Joseph Haydn: Quartet in D Major, Op. 50, No. 6, "The Frog," Minvetto Allegretto and Finale Allegro con spirito. The quartet members are Anne Nagosky, violin; Marrie Arnold, violin; Ana Ruesink, viola; Kathy Cherbas, cello.

Session BB. Engineering Acoustics I, Psychological and Physiological Acoustics VII, and Speech Communication VI: Engineering Approaches to Hearing Problems

Mead Killion, Chairman
Etymotic Research, Inc., 61 Martin Lane, Elk Grove Village, Illinois 60007

Chairman's Introduction—8:15

Invited Papers

8:20

BB1. An overview of the functions of the ear. Murray B. Sachs (Department of Biomedical Engineering and Center for Hearing Sciences, Johns Hopkins University School of Medicine, Baltimore, MD 21205)

The function of the peripheral auditory system (external, middle, and inner ears) is to produce a representation of sound across the population of auditory-nerve fibers. This representation is processed by the brain to produce appropriate behaviors (e.g., perception, movement, etc.). This talk will focus on the representation of the spectral features of speech sounds in the auditory nerve. Emphasis will be placed on those aspects of the encoding in the auditory nerve that allow the representation to be robust over a large range of sound levels and in the presence of background noise. The transformations of this representation by populations of cochlear nucleus cells will be discussed. [Work supported by grants from NINCDS.]

8:42

BB2. Digital hearing aids. Harry Levitt (Graduate School, City University of New York, 33 West 42nd Street, New York, NY 10036)

Digital hearing aids offer many advantages over conventional hearing aids. These include programmability, communication with and control by a host computer, and the potential for implementing advanced signal processing techniques for reducing background noise and for enhancing speech intelligibility. Three types of experimental digital hearing aids have been developed: (1) a quasidigital system in which the audio signals remain in analog form but are controlled by digital means, (2) a digital filtering system under microprocessor control, and (3) real-time simulation of a prescribed hearing aid on a high-speed, general-purpose computer. Recent experiments using digital hearing aids have evaluated new methods of hearing-aid prescription, techniques for processing speech to improve intelligibility for the hearing impaired, methods of noise reduction, and new approaches to the measurement and specification of hearing-aid characteristics. Existing experimental digital hearing aids are relatively large in size, the smallest being a body-worn unit. Current engineering efforts are directed towards developing a unit small enough to be worn on or in the ear.

9:04

BB3. The middle ear from the standpoint of the surgeon and the acoustician. Richard L. Goode (VA Medical Center, Palo Alto, CA 94304) and Mead C. Killion (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007)

Recent data obtained from fresh human temporal bones [Gyo *et al.*, Ann. Otol. Rhinol. Laryngol. (1987)] have provided previously unavailable information regarding the likely effect on the middle ear transmission resulting from surgical modification. The possibility of improving a low-grade ear (from the middle ear transmission standpoint) into a "golden ear" will be discussed. Calculations based on modifications to the analog external-plus-middle-ear model of Killion and Clemis [J. Acoust. Soc. Am. Suppl. 1 **69**, S44 (1981)] will be presented.

9:26

BB4. Cochlear implants: What are they and what do they accomplish? Donald K. Eddington (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

A large portion of the profoundly deaf are not able to benefit significantly from conventional hearing aids. Even with the tremendous gain provided by high-powered aids, the peripheral auditory system of these impaired individuals is not able to transform the intense sound input to patterns of neural activity that the brain can interpret. Cochlear implants are devices that convert sound to electrical stimuli and use electrodes placed in the cochlea to deliver these stimuli to the auditory nerve. The neural activity elicited by these stimuli is interpreted as sound percepts by the brain. The goal of these devices is to create patterns of neural activity that the brain can learn to interpret and thereby restore some measures of hearing to the deaf. These devices are now commercially available in several forms and can provide significant benefit to the profoundly deaf individual. This presentation will describe the rationale of cochlear implantation, the classes of devices now available, and the benefits provided to users of these devices.

9:48

BB5. Tactile aids for the deaf. Moise H. Goldstein, Jr. (Johns Hopkins University, Electrical Engineering Department, Baltimore, MD 21218)

In a recent article [C. E. Sherrick, J. Acoust. Soc. Am. **75**, 1325–1342 (1984)], some noteworthy questions about tactile aids were posed. Among them were: "(i) What are the processing capacities of the skin?, (ii) what form of transducer system will provide a reliable and efficient display to the skin?, (iii) if not all speech features can be handled by the substitute channel, which are the important ones to emphasize?, and (iv) how should the target population be defined?" These questions will be considered in context of the present day. Several companies now sell tactile aids for amounts close to the cost of a quality hearing aid. The characteristics and marketing of these commercial aids will be indicated. The target population of children versus adults and prelingually deaf versus postlingually deaf will receive special consideration. At present, one may question whether tactile aids are effective sensory substitutes for audition; nevertheless, it is becoming clear that they can have important benefits. [Work supported in part by NSF and NIH.]

10:10–10:15

Break

Contributed Papers

10:15

BB6. Coupled pendulum model for the middle ear. Ahmed M. Nassef, R. D. Finch (Department of Mechanical Engineering, University of Houston, Houston, TX 77004), and L. Gray (Department of Otolaryngology—Head and Neck Surgery, University of Texas Medical School, Houston, TX 77030)

Models of the middle ear given in the literature (piston and lever, membrane and lever, two piston and lever) were studied by deriving their transfer functions, and the values of the parameters best fitting an experimental transfer function were derived for each model. It was found that none of these models could fit the data with the same physically realizable components over the whole audible frequency range. Consequently, a new model was sought, based on an inspiration from Békésy's observation of rotational motion of the tympanum. That the middle ear acts as a two-degree-of-freedom system comprised of two coupled pendulums is suggested. Such systems have an inherent out-of-phase motion at high frequencies. That the malleus and part of the tympanum could comprise one rotating unit and that the incus and stapes could comprise the other is also suggested. The incudo-malleol joint could then be represented by the coupling and fulcrum in the model. Such a model has been found to meet the objective of fitting the transfer function over the entire frequency range. It is possible that such models can have application in prosthesis design.

10:30

BB7. On the measurement and calibration of ear impedance and its relation to the frequency of test signal. Yu-An Rao, Zhian Liang (Institute of Physiology, Academy of Sciences, Shanghai, People's Republic of China), and Edward C. Carterette (Department of Psychology, University of California, Los Angeles, CA 90024)

Experimental measurements and comparison of calibrated equivalent volumes of acoustic compliance were carried out with a probe set under different lengths of tube neck and diameters of coupling mouth. The relationship between equivalent volumes of acoustic compliance was a monotonic function of frequency for test frequencies below 500 Hz, but changed to a V-shaped function of frequency as test frequency was raised higher than 800 Hz. In cases where neck length ranged between 0 to 5 mm and the coupling diameter range between 1 and 2 mm, the maximal calibration error caused by the interaction of coupling mouth diameter and tube-neck length reached 0.4 ml for low frequencies; the maximal calibration error was much larger for high frequencies. In addition, comparisons of five different calibration cavities were carried out on, for example, Madsen and Chinese laboratory and clinical types. On the basis of these measurements on different cavities, it is proposed to take as a standard the value of 2 mm for both the tube-neck length and the coupling diameter.

10:45

BB8. Interpretation of amplitude compression in terms of the modulation transfer function. R. Plomp (Department of Otolaryngology, Free University Hospital, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands)

The successful application of the speech transmission index for predicting the combined effects of reverberation and noise on the intelligibility of speech [e.g., T. Houtgast and H. J. M. Steeneken, J. Acoust. Soc. Am. 77, 1069–1077 (1985)] demonstrates that the modulation transfer function (MTF) is an appropriate measure of the consequences of temporal modifications of the speech signal on its perception. In view of this, it makes sense to describe also the effect of amplitude compression on speech intelligibility in terms of the MTF. A compression circuit, as applied in hearing aids, reduces the MTF from zero up to a critical modulation frequency of, typically, 10 Hz. This means that a substantial part of the range of modulation frequencies present in the speech signal and relevant for its intelligibility (0.4–20 Hz) is affected by the compression process. This can explain why amplitude compression with short time constants has a negative effect on the intelligibility of speech in noise. The results of other attempts to modify the MTF for better speech understanding will also be discussed.

11:00

BB9. Functional gain measurement of both amplitude and phase characteristics. Harry Levitt (Center for Research in Speech and Hearing Sciences, City University of New York, 33 West 42nd Street, New York, NY 10036)

THURSDAY AM

The principle underlying the measurement of functional gain in hearing aids is that of comparing the stimulus level at threshold for the test condition (e.g., the aided condition) with the threshold level for a reference condition (e.g., the unaided condition). This technique is extremely useful but is limited, in its present form, to the measurement of amplitude-frequency characteristics only. The technique has been extended to the measurement of phase-frequency characteristics by introducing a secondary reference, a sinusoidal signal delivered to the ear by an alternate route (e.g., by bone conduction). The signal generated by the system under test is adjusted in amplitude and phase so as to cancel the secondary reference signal. The cancellation is done subjectively with the test system *in situ*. The signal generated under the reference condition is canceled in the same way. A comparison of the two sets of adjustments needed for cancelation determines the functional gain of the system with respect to both amplitude and phase.

11:15

BB10. Separation of simultaneous voices. Richard J. Stubbs and Quentin Summerfield (M. R. C. Institute of Hearing Research, University of Nottingham, University Park, Nottingham NG7 2RD, United Kingdom)

People with sensorineural hearing impairments find the task of separating speech from interfering noises particularly difficult. For this reason, the ideal hearing aid would selectively amplify a target voice while attenuating competing voices and other background noises. In exploring possible noise-reduction strategies for future aids, two algorithms for separating the voices of talkers who are speaking simultaneously, with periodic excitation at similar overall intensities, are evaluated. Both approaches are pitch based and exploit the regularity in the harmonic structure of voiced speech. The first involves attenuating the harmonics of the competing voice via a cepstrumlike representation. The second method is derived from the procedure for harmonic selection [T. W. Parsons, J. Acoust. Soc. Am. 60, 911–918 (1976)]. Perceptual evaluation of the two processing methods, in a test involving the separation of concurrent vowel pairs synthesized on fixed fundamental frequencies, has demonstrated an increase in performance for both normal-hearing and hearing-impaired subjects. The results of experiments involving the separation of simultaneous sentences containing a high proportion of voiced excitation and varying fundamental frequencies will also be discussed.

THURSDAY MORNING, 14 MAY 1987 — CANYON HALL, 8:30 A.M. TO 12:05 P.M.

Session CC. Musical Acoustics III: Acoustics of Pianos I

Gabriel Weinreich, Chairman
Randall Laboratory, University of Michigan, Ann Arbor, Michigan 48109

Chairman's Introduction—8:30

Invited Papers

8:35

CC1. Piano design factors—Their influence on tone and acoustical performance. Harold A. Conklin, Jr. (Box 1915, Dunedin, FL 34296)

Major design factors that affect significantly the tone and acoustical performance of the modern mechano-acoustic piano are reviewed and the influence of each on the performance of the instrument is briefly described or illustrated. Elements discussed include the string plate, the soundboard, the ribs, the bridges, the "scale," the strings, the hammers, the frame, the case, and the lid. Some directions are suggested for future work to improve performance. The presentation is from the perspective of the instrument designer.

9:05

CC2. Piano stringing-scale design. Albert E. Sanderson (Inventronics, Inc., 171 Lincoln Street, Lowell, MA 01852)

Stringing-scale design in pianos comprises the specification of the speaking length of each note, the number of strings per note (unichord, bichord, or trichord), the diameters for all plain-wire notes, and the diameter of the core and diameter and length of the copper winding for all loaded strings. These factors control the inharmonicity and tension of each string, and control of both is essential for excellent volume, tone quality, and balance of sound from bass to treble in a fine grand piano. While there is no accounting for taste, there is general agreement on which pianos sound best. Design rules have been derived based on these pianos, as well as on a large number of poor pianos, that reliably predict what ranges of the above design parameters will "sound good." In general, the larger the piano, the more tension and the less inharmonicity it must have for excellent tone quality. Inharmonicity must also be carefully controlled to achieve perfect tunability. This is a rare quality that allows an instrument to be tuned so well that every musical interval (when played in ascending sequential semitones) increases smoothly and evenly in beat rate.

9:35

CC3. From touch to string vibrations—The initial course of the piano tone. Anders Askenfelt and Erik Jansson (Department of Speech Communication and Music Acoustics, Royal Institute of Technology KTH, P.O. Box 70014, S-100 44 Stockholm, Sweden)

In a pilot study [J. Acoust. Soc. Am. Suppl. 1 **71**, S92 (1982)], measurements of key velocity, hammer acceleration, and string vibrations were reported. The study also covered the timing of the parts in the action and contact durations between hammer and string. The project was now continued by verifying measurements of key motion and string vibrations by means of a novel optical detector. In addition, the motion of the hammer has been recorded simultaneously with key motion under different playing conditions (pianist, nonpianist, pendulum). The effects of adjustment of the action on timing, as well as the effects of weight, shape, and softness (intonation) of the hammer on the string waveform and spectrum, have also been investigated. The methods and the results of the measurements will be presented and discussed.

10:05

CC4. Input impedance and sound radiation of the piano sound board. Klaus Wogram (Physikalisch-Technische Bundesanstalt, 3300 Braunschweig, Federal Republic of Germany)

The soundboard of the piano acts as the signal converter from the string vibration to the airborne sound. The type of mounting of the soundboard on the back, as well as the ribs, influences both the sound radiation and the input impedance that acts as a load for the strings as vibration generators. Furthermore, the iron plate affects the effectiveness of the soundboard. Extensive measurements have shown that mass loading by the ribs is in general too large to compensate for the unisotropism in its modulus of elasticity. This causes an additional damping and sometimes an increase of the decay rate of the piano sound.

Contributed Papers

10:35

CC5. Longitudinal vibrations in piano strings. Michael Podlesak and Anthony R. Lee (Physics Department, La Trobe University, Bundoora, Victoria-3083, Australia)

The percussive excitation of a piano string by a hammer is expected to produce longitudinal string vibration components in addition to the desired transverse ones [e.g., Yanagisawa *et al.*, J. Acoust. Soc. Jpn. **33**, 412–416 (1977)]. However, so far little quantitative information exists in current acoustics literature on the contribution of such longitudinal components to the radiated sound of a piano. Our investigations revealed the presence of intial components of radiated sound generated by the longitudinal string vibration mechanism. These components, coined as the *precursive sound*, appear distinctly ahead of the main body of sound that is almost entirely due to transverse string vibration. Our experimental data on the longitudinal modes in piano strings indicate that the most apparent contribution to the radiated sound occurs in the attack transient of tones in the low bass register. For the lowest bass tones, the sound-pressure level (SPL) of the *precursive sound* may be only 10 to 20 dB below the overall peak SPL and the decay rate of the order of 100 dB/s. [Work supported by A.R.G.S. and C.T.E.C.]

10:50

CC6. Vibrations of a piano soundboard: Modal analysis and finite element analysis. Joseph Kindel and I-Chih Wang (3903 Hemphill Way, Cincinnati, OH 45236)

Modal analysis measurements have been made on two concert grand piano soundboards and a finite element model to describe the vibrational characteristics of these soundboards has been developed in order to explore the possibility of using both of these techniques in the designing of new soundboards. To provide an experimental basis for the model, modal analysis measurements were made on two piano soundboards, each mounted in a rim. Then a finite element model of the soundboard was constructed. The resonant frequencies and mode shapes of the first few modes of the finite element model match very well with those found experimentally using modal analysis. The correspondence between the modal analysis and finite element analysis results is very good and these techniques will be of value in designing new piano soundboards. [Work supported in part by Baldwin Piano and Organ Co., Cincinnati, Ohio.]

11:05

CC7. Analysis and synthesis of piano tone. Astrid Reinholdt (Laboratory of Acoustics, Technical University of Trondheim, N-7034 Trondheim, Norway), Erik Jansson, and Anders Askenfelt (Department of Speech Communication and Music Acoustics, Royal Institute of Technology KTH, P.O. Box 70014, S-100 44 Stockholm, Sweden)

Signals from a grand piano, consisting of the string velocity and the sound in the room, were recorded. The signals were analyzed with regard to waveform and spectral development in time. The analysis, as well as a listening to the recording, revealed a prominent initial thump in the room signal that preceded the tonal string sound. This attack sound was not present in the string velocity signal. Piano tones were synthesized by modeling the analysis results in an additive computer synthesis. Important features in the synthesized piano tone, such as spectral envelopes, initial thump, inharmonicity in strings, and multiple stringing, are illustrated by sound examples.

11:20

CC8. Vocal register change: A study of perceptual and acoustic isomorphism. Anat Keidar (Department of Otolaryngology, Washington University Medical Center, Jewish Hospital, 4910 Forest Park Boulevard, Suite 212, St. Louis, MO 63108)

This study focuses on the perceptual nature of chest and falsetto registers, and on the degree of correspondence between perception and several acoustic measures. Fifteen target notes, ranging from G#3 to A#4, were sung by a male and a female subject in the context of ascending and descending sequences on the vowels /i/ and /a/. Register transitions were elicited by setting strict constraints on production, and by minimizing auditory feedback. Segments of 1-s duration were extracted from the target notes, digitized, acoustically analyzed, and perceptually judged by ten trained listeners. Multidimensional scaling and hierarchical clustering analyses were utilized to capture the dimensionality and the internal structure of perceptual data sets derived from pairwise similarity ratings. Similar analyses of the acoustic data sets provided a means for evaluating the congruence of each acoustic variable and perception. Optimal spatial representation of the perceptual data required no more than two orthogonal dimensions, with the quality attribute represented by the dominant dimension. The acoustic variables which exhibited the highest degree of

THURSDAY AM

isomorphism with perceived registers were characterized by differences in frequency spectra obtained from a set of 1-oct bandpass filters, and by $F_0/$ harmonic ratio. The representation of pitch differences was reflected only in the internal ordering of the stimuli within register, but did not affect the perceptual discontinuity between registers.

11:35

CC9. Progress toward more intuitive programming of FM synthesizers. G. L. Gibian, D. R. Clements, E. N. Harnden (Physics Department, American University, Washington, DC 20016), and D. L. Bort (Johns Hopkins Applied Physics Laboratory, Johns Hopkins Road, Laurel, MD 20707)

As was reported last year, work has been done to make the programming of new tone colors on FM synthesizers more intuitive for composers and musicians. The logarithmic relations between modulator, carrier, and envelope level parameters on commercially available instruments and corresponding index of modulation and output amplitudes in standard FM synthesis equations have been empirically determined. The relations have been incorporated into a tutorial computer program that displays the waveform and corresponding spectrum after synthesis parameters are entered in either form. With the program, users can gain familiarity with the spectra of FM-synthesized sounds, which can be quite complicated for synthesizers using four or six oscillators in various configurations. Work is currently being done to calibrate feedback and envelope rate parameters and to develop software to go in the opposite direction, determining FM parameters to approximate spectra with complicated and time-varying shapes. Ultimately, such software should choose the configuration of oscillators as well as other parameters to create the perceptually closest approximation to the desired spectral characteristics.

11:50

CC10. Absolute pitch: A methodological fallacy. Rudolf A. Rasch (Department of Phonetics, University of Utrecht, Trans 14, 3512 JK Utrecht, The Netherlands)

The research on absolute pitch (AP) has already celebrated its first centenary. So far, researchers (and the public) have divided listeners into AP possessors and non-AP possessors. Analyses of the tasks included in AP performances show that it is unsuitable to speak about AP or RP (relative pitch) without qualification. Certainly there are large individual differences in the capacity of naming the pitch of a tonal sound presented without prior reference. But responses are bound to be (more or less) normally divided, with the standard deviation as measure of performance. In such a view, the decision of whether or not a person is an AP possessor depends on the category width of the labels, which means that the distinction between AP and non-AP is relative, not absolute.

THURSDAY MORNING, 14 MAY 1987 — MT. McKINLEY, 8:30 TO 11:55 A.M.

Session DD. Underwater Acoustics V: Matched Field Signal Processing—Part I

Orest Diachok, Chairman
Naval Research Laboratory, Washington, DC 20375

Chairman's Introduction—8:30

Invited Papers

8:35

DD1. Use of matched fields and other methods to locate sound sources or reflectors in the ocean. Homer P. Bucker (Code 541B, Naval Ocean Systems Center, San Diego, CA 92152) and Philip W. Schey (Computer Sciences Corporation, 4045 Hancock Street, San Diego, CA 92110)

A fundamental difference in locating a sound source underwater and a visual source in our everyday experience is that the sound waves are strongly bent by the variation of sound speed as a function of depth and range in the ocean. With the advent of inexpensive computing machines, it is now possible to begin to unscramble the refraction and reflection effects of underwater sound propagation to "see" the location of underwater sources and reflectors of sound. This paper reviews earlier theoretical and experimental work in matched field processing where the typical problem has been to determine the range and depth of an underwater source using narrow-band signals received at a vertical line array. In future years, it is clear that these methods will be expanded to cover wideband signals and reflective as well as active sources. A method will be discussed where correlations are made between calculated and measured plots of sound pressure as a function of frequency and time. [Work supported by NOSC IR Program.]

9:00

DD2. An evaluation of matched field processor performance with environmental mismatch. Ronald L. Dicus (Code 5120, Naval Research Laboratory, Washington, DC 20375)

The matched field processor is enjoying a technical renaissance because of increased computer computation speeds and environmental modeling capability. Its performance, however, is degraded by error or mismatch in waveguide steering vectors (theoretical fields) and by ambient noise. The effects of error in sound velocity profile, water depth, array depth, and sensor positions will be discussed in relation to their effect on source location bias, peak height, and peak width as a function of mismatch strength, frequency, range, and array length. Noise effects are not considered. To gain insight, the processor output was investigated for a simple multipath signal environment in which acoustic energy travels along straight line trajectories but with allowance for different effective wave speeds. At sufficient ranges it can be shown that the processor is equivalent to a linear sum of focused beamformers, each steered to a particular multipath. Error in the waveguide steering vectors has the effect of differentially rotating theoretical beams relative to true field beams, resulting in localization bias and peak height reduction. Ambiguous peaks are caused by rotation of a theoretical beam for one multipath onto the true field beam of another multipath. These effects will be discussed with numerical examples.

9:25

DD3. Matched field processing: A detection and estimation theory overview. A. B. Baggeroer (MIT, Cambridge, MA 02139), W. A. Kuperman (Naval Research Laboratory, Washington, DC 20375), and H. Schmidt (SACLANT ASW Centre, I-19026, La Spezia, Italy)

Matched field processing in weak signal/noise situations requires maximum exploitation of the physics of both the signal and noise structure as well as optimum methods for the signal processing. In recent years, numerical models for the physics have advanced to the point where many of the detection and estimation theory results available in the signal processing literature can now be used for many acoustic environments. Fundamentally, matched field processing is known as the detection/estimation in colored, or correlated, noise problem in the signal processing literature. As such, ambiguity functions, Cramer–Rao bounds, and spectral measures of performance can be derived from a basic, well-known theory. Generalizations to high-resolution algorithms and environments with partial coherence are also suggested. Finally, results for the less studied problem of matched field processing performance using estimated covariance matrices can be addressed.

9:50

DD4. Source location and identification by backpropagation in the time domain. Robert P. Porter and Jei Shuan Chen (Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

Generalized backpropagation in the time domain has been applied to seismic inversion and is now being extended to the source location and identification problem in underwater acoustics. The method is based on a discrete implementation of the Huygens–Kirchhoff integral and is applicable to a wide range of wave-front migration problems including underwater acoustics. The computation is simplified by using a few nearest neighbors to calculate the field along a nearby coplanar surface. A brief discussion of generalized backpropagation in both the frequency and the time domains will be presented. The method will be illustrated by showing reconstructions of some of the numerous spread-source geometries that have been studied. Some implications of the use of this method for locating and identifying sources in the ocean will be discussed.

10:15

DD5. Signal field matching in range, depth, and bearing. John M. Ozard (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

Advances in propagation modeling have enabled increasingly accurate calculations of the acoustic field for situations where fairly complete knowledge of the acoustic environment is available. To take advantage of the improved propagation models, various matched field signal processing schemes have been developed. Because of the large number of possible combinations of range, depth, and bearing for a moving acoustic source, it is desirable to speed up the matching process and reduce the search region. This can be achieved by using an initial coarse global grid search followed by a fine optimization of source position to produce field matches. Further gains can be obtained by predicting the next source position so that subsequent searches can be restricted to small regions around the predicted positions. In this presentation, the essence of the normal mode propagation model and suitable processing and prediction schemes are described and applied to a shallow water environment. This is followed by illustrations of performance for a generalized beamformer and for an orthogonal technique under a variety of environmental conditions, source parameters, and array configurations.

Contributed Papers

10:40

DD6. Passive ranging as an inverse problem. E. J. Sullivan (SACLANT ASW Research Centre, APO New York, NY 09019)

A passive ranging technique is presented which uses SNAP, the SACLANT Centre's normal-mode model. A set of linear equations are obtained that allow a maximum likelihood estimate of the range of an acoustic source to be directly computed using any desired combination of modes. This is in contrast to the matched-field techniques that use a search over forward solutions, each of which assumes a particular source location. The sensitivity of the technique to errors in the assumed sound velocity profile and tilting of the 16λ vertical receiving array is studied. Also, the effect of both white and colored noise is investigated. Results indicate that, for the scenario considered (Summer profile in the Mediterranean with a water depth of 130 m and source ranges of 10 and 25 km), the method can be quite robust to sound velocity profile errors on the order of several m/s and array tilts on the order of 1.5°. Solutions were obtained with input signal-to-noise ratios as low as -17 dB.

10:55

DD7. A simple method of range and depth estimation. T. C. Yang (Naval Research Laboratory, Washington, DC 20375)

The range and depth of an acoustic source in the ocean can be determined by decomposing array data and beamforming on the mode amplitudes. In particular, because the mode amplitudes are proportional to $\exp(-ik_i r_s)$ as a function of source range, the product of the mode amplitudes with the steering vector V [where $V_i = \exp(ik_i r)$] is maximum for the true source range ($r = r_s$). Accurate range and depth estimation with this approach, however, requires reliable estimates of the mode amplitudes—often difficult to obtain. In this paper, an eigenvector decomposition technique is used to extract the mode amplitudes for data received on a long (1-km) vertical array. Range and depth are successfully estimated both for simulated data and for data from the 1982 Fram IV Experiment in the Arctic Ocean. The effect of the number of modes on the estimates is also illustrated.

11:10

DD8. Simulation of matched field processing for convergence zones in a strong bottom field. David F. Gordon (Naval Ocean Systems Center, Code 711, San Diego, CA 92152-5000)

Below 30 Hz, the convergence zone energy can be smaller than the bottom reflected energy. Matched field simulations using a normal mode program give narrow, irregular correlation peaks for target vectors taken from the complete field. Target vectors computed using only modes representing convergence zone energy are then processed against the complete acoustic field. A much broader, smoother correlation function is obtained. Target vectors are then constructed that are equivalent to delay and sum beamforming at specified angles. These vectors give correlation functions that resemble those for convergence zone energy, and illustrate the relationship between matched field processing and beamforming.

11:25

DD9. Range-depth ambiguity surfaces for vertical arrays in shallow-water environments. Richard Heitmeyer and Rachel Hamson (SACLANT ASW Research Centre, APO, New York, NY 09019)

This paper describes an analytical study of range-depth ambiguity surfaces obtained on a vertical array in shallow water through the spatial correlation of the incident acoustic field with a replica of that field. The results indicate that a high-quality ambiguity surface (narrow main lobe-low sidelobe levels) can be obtained provided that the incident acoustic field is formed from a significant number of "effective" modes and that the array spans a large fraction of the water column. Changes in the acoustic environment that reduce the effective number of modes diminish the quality of the ambiguity surface. Thus the quality is reduced by either a decrease in the water depth or the source frequency or by an increase in either the source range or the bottom loss. Furthermore, although the array length is a key parameter, hydrophone spacings over twice that required to satisfy a Nyquist criteria (water depth/number modes) can be used without substantial degradation. Finally, errors in the bottom parameters, which produce errors in the replica field, shift the main-lobe peak, whereas errors in the sound-speed profile can degrade the ambiguity surface to the point where the main-lobe peak is indistinguishable from the sidelobe peaks.

11:40

DD10. Experimental determination of modal depth functions from covariance matrix eigenfunction analysis. Stephen N. Wolf (Code 5160, Naval Research Laboratory, Washington, DC 20375 and Defence Research Establishment Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada)

A low-frequency sound field in a shallow-water duct is usually represented by a set of discrete normal modes, each of which is characterized by a unique vertical pressure amplitude depth function. In many applications, such as matched-field signal processing, it is important to have accurate determinations of these depth functions, which are usually calculated from normal-mode models. By using a densely populated vertical array spanning the water column and a large time-bandwidth test signal, in some circumstances the normal-mode functions of the duct may easily be obtained by diagonalizing the cross-spectral density function of the received signal. The condition required for using this technique is that, over the bandwidth used, the fields of individual modes must be mutually incoherent so that the cross-spectral density matrix is the sum of individual modal diads, making the modal depth functions the same as the matrix eigenfunctions. Experimentally determined mode functions of orders 1 through 4 are compared with theoretical results.

THURSDAY MORNING, 14 MAY 1987 REGENCY BALLROOM A & B, 8:45 TO 11:45 A.M.

Session EE. Speech Communication VII: Linguistics, Phonetics, and Women's and Children's Voices (Poster Session)

Paul A. Luce, Chairman

Department of Psychology, Indiana University, Bloomington, Indiana 47405

All posters will be displayed from 8:45 until 11:45 a.m. To allow all contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:45–10:15 a.m., and contributors of even-numbered papers will be at their posters from 10:15–11:45 a.m.

Contributed Papers

EE1. The phonetic relevance of temporal decomposition. Stephen M. Marcus (AT&T Bell Laboratories, Murray Hill, NJ 07974 and Institute for Perception Research—IPO, Eindhoven, The Netherlands) and Ann K. Syrdal (AT&T Bell Laboratories, Naperville, IL 60566)

A simplified view of speech production considers it to be the result of a sequence of gestures toward articulatory targets, these gestures overlapping in time and thus resulting in spectral transitions in the acoustic speech signal. Temporal decomposition is an acoustic analysis technique which, without using any detailed phonetic knowledge, decomposes a speech signal into a sequence of spectral target vectors, each associated with a time varying target function. [B. S. Atal. Proc. ICASSP **83**, 81–84 (1983); S. M. Marcus and R. A. J. M. van Lieshout, IPO Annual Progress Report, **19**, 25–31 (1984)]. This report will further evaluate relationships between the acoustic characteristics of phonetic units and the target vectors and functions resulting from temporal decomposition. Comparisons will be made between the temporal alignment of phonetic segments and target functions. The effect on target function duration of systematic contextural effects known to affect segment durations, such as final consonant voicing on preceding vowel duration, will be explored. Spectral properties characteristic of representative vowels and consonants will be compared with those of the corresponding target vectors, and some estimates of the stability of these values will be given.

EE2. Linguolabials. Ian Maddieson (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

A number of Austronesian languages spoken in Vanuatu have a series of linguolabial consonants (i.e., stops, nasals, and fricatives produced by articulation of the tongue blade with the upper lip). This paper presents the first detailed study of the production and acoustic nature of linguolabials based on data from three of these languages. These sounds may be produced with varying degrees of retraction of the upper lip and protrusion of the tongue, but all involve a vocal tract configuration with a narrow cross-sectional area in the front region. In linguolabial continuants the second resonance of the vocal tract is higher than it is in labials, and the burst of the linguolabial stop has greater energy at high frequency than a labial stop. These segments thus have some similarities with dental segments. Linguolabials are perceptually difficult to distinguish from dental or labial sounds, depending on phonetic context. The occurrence of contrast between labials, linguolabials, and dentals in these languages indicates that quite subtle details in the spectrum must be adequate to signal differences in place of articulation. [Work sponsored by NSF.]

EE3. Flapping of apical stops in American English. Fares Mitleb (Department of English, Yarmouk University, Irbid, Jordan)

One fundamental assumption in all theories of phonology has always been that neutralization rules have the power to obliterate the contrast between segments in certain environments. For example, American English is well known for its neutralization of intervocalic, poststressed /t/ and /d/ to an apical "flap" [r]. However, acoustic analysis of five pairs of English words produced by five native speakers of American English revealed that flapping of apical stops is a near-neutralization process. Furthermore, in a listening test, American listeners were able to distinguish underlying /t/ and /d/ with about 69% accuracy—significantly better than chance. This suggests that segmental feature analysis oversimplifies the speech production and perception processes, since it states in advance the facts to be accounted for in phonology. Results of this study seem to call into question the current formulations of neutralization rules and related concepts. The findings will be interpreted in terms of their consequences for the presence of temporal implementation rules. [Research supported by Yarmouk University, Jordan.]

EE4. Effects of pragmatics on phonetic variables. R. Port and P. Crawford (Indiana University, Bloomington, IN 47405)

It has been shown that the supposed neutralization of German minimal pairs like *bunt* and *Bund* is incomplete [J. Phonet. (1986)] since small but consistent differences can be found in phonetic variables. This permits listeners to guess with better-than-chance accuracy. But, since the distinction is very small, a question arose as to whether speakers can control the amount of difference they can produce between the two categories. In an experiment speakers were given a variety of tasks involving pronunciation of a small set of minimal pairs of the type *bunt–Bund*. First, they read the words within a long list of sentences so that their attention was diverted away from the test words. Then carrier sentences with the words in adjacent sentences were read, and finally German sentences comparable to, "No, I didn't say *bunt*, I said *Bund*," were read. Analysis of temporal measurements from the words showed that speakers did produce more distinctive pronunciations in contexts where their attention was drawn to the contrast, and even more when trying to distinguish the pronunciations for a listener. This is evidence of graded phonetic control even when speakers have faint awareness at most that a distinction is made. [Research supported by NIH, HD12511 and NSF.]

EE5. Effects of voicing, place, and vowel context on VOT for French and English stops. Bernard L. Rochet, Terrance M. Nearey, and Murray J. Munro (Departments of Linguistics and Romance Languages, University of Alberta, Edmonton, Alberta T6G 2E7, Canada)

Preliminary results in a study of VOT in the production of French initial stops (recorded from eight native speakers) indicate a significant interaction of voicing state, place of articulation, and vowel identity in /CVk/syllables. While some of the trends observed are similar (e.g., VOT for high vowels is generally longer for that of non-high vowels), the exact pattern of interactions does not agree with that found for American English [R. F. Port and R. Rutonno, J. Acoust. Soc. Am. **66**, 654–662 (1979)]. Results from a larger-scale measurement study comparing VOT production in French and Canadian English will be discussed. Interaction effects that depend explicitly on categorical (as opposed to purely acoustic) context have been shown to be reliable in VOT production experiments [Port and Rutonno (1979)]. Results of cross-language perceptual experiments (using VOT continua in which vocalic context and place of articulation cues are varied as parameters) designed to test the perceptual reliablity of such effects will be discussed. [Work supported by SSHRC.]

EE6. The "effective second formant" F_2': A parameter for the classification of French front vowels. Marios Mantakas, Pierre Escudier, and Jean-Luc Schwartz (Institut de la Communication Parlée, Unité associé au C.N.R.S. No. 368, ENSERG, INPG, 46 avenue Félix-Viallet, 38031 Grenoble Cedex, France)

The "effective second formant" F_2' was analytically tested as a cue for rounding opposition in French front vowels (/i/–/y/ and /e/–/ø/ opposition). For this purpose, a corpus of these four vowels introduced into contexts maximizing and minimizing the rounding opposition [C. Abry, L-J. Boë, P. Corsi, R. Descout, M. Gentil, and P. Graillot, *Labialité et Phonétique* (Université des Langues et des Lettres, Grenoble, France, 1980)] was acoustically studied. The first four formant frequencies and levels were measured. The F_2' of the analyzed vowels was next estimated with a model based on psychophysical and vowel perception hypotheses and using the frequencies and levels of F_2, F_3, and F_4 [M. Mantakas, J-L. Schwartz, and P. Escudier, *Actes des 15e Journées d'études sur la Parole* (GALF, Aix-en-Provence, France, 1986), pp. 157–161]. The effectiveness of F_2' as a classification parameter was furthermore compared with that of other acoustic cues proposed for French. [Work supported by CNET, France.]

THURSDAY AM

EE7. Discrimination of duration ratios in Estonian and English. Robert Allen Fox (Speech and Hearing Science, Ohio State University, 324 Derby Hall, Columbus, OH 43210) and Ilse Lehiste (Linguistics, Ohio State University, 204 Cunz Hall, Columbus, OH 43210)

Estonian has been described as having a three-way quantity distinction among disyllabic words based upon whether the ratio of the durations of syllable 1 to syllable 2 is approximately 2:3, 3:2, or 2:1. The present study was designed to examine the ability of listeners to discriminate among such duration ratios independent of other phonetic factors, such as fundamental frequency or segmental variations. Pairs of noise bursts (with 10 ms rise/fall times) whose durations were in the ratios of 1:2, 2:3, 3:2, and 2:1, were constructed. In half of these noise sequences the combined duration of noise 1 + noise 2 was 350 ms; in the other half the combined duration was 450 ms. Subjects were presented with two such sequences and were required to respond whether the duration ratios in the two sequences were the same or different. Equal numbers of sames and differents were randomly presented to 56 native listeners (and five trained phoneticians). Twenty-eight listeners were native Estonians (whose data were collected in Tallinn, Estonia) and 28 listeners were native American English speakers (whose data collected in Columbus, OH). Obtained results show that both groups of listeners clearly recognized only two contrastive patterns: 1:2 and 2:3 vs 3:2 and 2:1. In addition, overall sequence duration had a significant effect upon duration ratio discriminations. These results will be discussed in light of possible language group difference, phonetic training differences, and in terms of psychoacoustic limits on the contrastive use of duration in languages.

EE8. Effects of consonant length on vowel-to-vowel coarticulation in Japanese. Caroline L. Smith (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Vowel-to-vowel coarticulation across consonants can be seen as evidence of overlap between the articulatory gestures of the vowels. Further evidence for the organization of the gestures can be found in contextual variation in the durations of the vowels. This acoustic study of Japanese examines the effects of geminate consonants on the durations and the $F2$ values of the surrounding vowels in VCV utterances. Because of the syllable structure of the language, geminate consonants in Japanese might be expected not to close the preceding syllable, eliminating this potential source of vowel shortening. In agreement with this hypothesis, vowels did not shorten before geminate consonants (in fact they lengthened), and the second vowel affected the formants of the first vowel less across geminate than across single consonants. [Work supported by NSF.]

EE9. The relationship between vowel variation and vowel-to-vowel coarticulation across languages. Brian McAdams (Department of Linguistics, Ohio State University, Columbus, OH 43210)

The amount of coarticulatory vowel variation within a language may be dependent on the size and distribution of the vowel inventory within the vowel formant space [P. A. Keating and M. K. Huffman, Phonetica **41**, 191–207 (1984)]. Similarly, it has been suggested that the extent of vowel-to-vowel coarticulation in a language varies in proportion to the degree of vowel variation within the language [S. Y. Manuel and R. A. Krakow, Haskins Status Report No. 77/78, 69–78 (1984)]. Manuel and Krakow supported this idea with evidence comparing Swahili and Shona (5-vowel systems) to English (12-vowel system). The present study compares vowel-to-vowel anticipatory and carryover coarticulation in Japanese and Hausa (10-vowel systems, with different vowel distributions) to German (14-vowel system). Three speakers of each language read bVbV "words" in lists and in frame sentences. The effects of phonemic vowel length and phrasal context on coarticulatory vowel distribution are also considered.

EE10. A cross-language study of vowel spaces and interference. Allard Jongman and Marios Fourakis (Central Institute for the Deaf, 818 South Euclid, St. Louis, MO 63110)

This is a study of the spectral characteristics of the vowels of German and modern Greek in relation to those of American English. Two native speakers of German produced 18 words containing the short and long vowels of German, and two native speakers of Greek produced five words containing the five pure vowels of Greek. All speakers also produced nine English words containing the nine simple vowels of English. Vowel target zones for German, Greek, and foreign-accented English are established within the auditory perceptual theory [J. D. Miller, J. Acoust. Soc. Am. Suppl. 1 **76**, S79 (1984)] and compared to the target zones of native speakers of English. A metric of the distance between the foreign, foreign-accented English, and native English target zones will be developed. Perception experiments are planned to obtain both identification scores and paired-comparison ratings of foreign accentedness. The positions of vowels in both the native and acquired language will be correlated with these ratings to determine if an objective measure of foreign accent and phonetic interference can be derived.

EE11. Perception of Persian uvular and velar stop consonants by speakers of American English. Linda Polka (Department of Psychology, University of South Florida, Tampa, FL 33620)

Recent cross-language studies by Werker and Tees [J. Acoust. Soc. Am. **75**, 1866–1878 (1984)] describe difficulties in English speakers' perception of a Thompson glottalized uvular-velar stop contrast. However, the acoustic information that differentiated the contrasting phones was difficult to specify. The present study extends the examination of non-English place contrasts by investigating native English speakers' ability to perceive a nonglottalized uvular–velar contrast in Persian (Farsi). Multiple instances of velar and uvular stop consonants produced in several VCV contexts by a native speaker of Farsi were presented. Correct identification of the medial consonant varied systematically as a function of vowel context from very few errors to near change performance. Acoustic analysis revealed systematic $F2$ and $F3$ formant transition differences between uvular and velar consonants produced in the same vowel context. Variations in voicing offset (murmur) were also observed in several vowel contexts. These results further our understanding of the role of linguistic experience in the perception of phonetic categories. [Research supported by NINCDS.]

EE12. Vowel duration in English word and sentence patterns as spoken by non-native speakers. Joann Fokes (School of Hearing and Speech Sciences, Ohio University, Athens, OH 45701) and Zinny Bond (Department of Linguistics, Ohio University, Athens, OH 45701)

Non-native speakers from five different language backgrounds (Chinese, Hausa, Japanese, Persian, and Spanish) and three native American English speakers produced tokens of two-syllable words (complete), three-syllable words (completion), and four-syllable words (competition) in isolation and in sentence context. Vowel duration, a correlate of stress, was measured in the first two syllables in words from spectrograms. The Americans used very short vowels in the reduced syllables and shortened stressed vowels in suffixed words and in sentence context. The primary difficulties of the non-native speakers were vowel reduction in unstressed syllables and the stress patterns of four-syllable words.

EE13. Phonetic reality and phonological prediction: Stress clash and rhythm in English. Keith Johnson (Department of Linguistics, Ohio State University, Columbus, OH 43210) and Dorothy Evans (Department of Linguistics, University of Illinois, Urbana, IL 61801)

Cooper and Eady [J. Mem. Lang. **25**, 369–384 (1986)] recently tested some predictions of metrical phonology only to find a lack of empirical support for some important aspects of the theory. This report will present the results of a replication and revision of one of their experiments. The phonological prediction being tested in this experiment is that the rhythm of an utterance will be adjusted toward isochrony is cases of stress clash (adjacent stressed syllables). In a pilot study it was found that in normal productions of test sentences the findings (or rather nonfindings) of Cooper and Eady were replicated. However, when subjects were instructed to "speak up," there was a tendency toward rhythmic adjustment of the duration of words involved in the stress clash. In the study to be reported in this paper, subjects produced stress-clash sentence pairs once normally and once carefully (the instructions were "as if you were talking to someone who might have a hard time understanding you"). The results of the pilot experiment seem to indicate that speakers do in some circumstances adjust the rhythm of an utterance in response to stress clash. This pattern of results raises some questions about what linguistic "competence" is being described by the phonological theories tested.

EE14. Using metrical phonology to synthesize speech. William Reilly (Department of Linguistics, Indiana University, Bloomington, IN 47405)

In this experiment synthesized reiterant speech is compared to natural reiterant speech. Using multiple linear regression, equations are obtained which take grammatical information (e.g., syllable weight, word boundary, lexical stress) of natural speech syllables as input and produce as output information sufficient to predict the timing of the corresponding reiterant speech in nonphrase final position. A corpus of 35 phrases was spoken reiterantly by three speakers, using the syllable *sa* as the reiterant syllable. Measurements of the duration of the reiterant segments were made automatically from the digitized speech and constitute the final dependent variable. Each syllable is grammatically described as a string of binary variables determined from metrical phonology [Liberman and Prince (1977)]. After calculating the predicted segmental durations of a reiterant phrase, the original speech was modified to fit these predictions using the waveform editor of ILS. The resulting synthesized speech is then judged by comparing it to the original reiterant speech of the three speakers. The success of this experiment suggests that the same task might be performed for natural speech, and that the grammar is tied more closely to production and perceptual mechanisms than previously shown. [Work supported by NSF.]

EE15. The phonological domains of final lengthening. Mary E. Beckman (Department of Linguistics, Ohio State University, Columbus, OH 43210) and Jan Edwards (Hunter School of Health Sciences, Hunter College, 425 East 25th Street, New York, NY 10010)

A syllable at the end of a phrase is considerably longer than it would be phrase-internally. Similarly, a stressed monosyllable is generally longer than any identical syllable separated from the word's edge by one or more unstressed syllables. Two sets of experiments have been designed to determine the phonological domain of these effects. The first compared target syllables at various distances from the ends of words ([pɛ] in *pep, pepper, peppermint*; [pa] in *pop opposed, poppa posed*) in sentences where they either did or did not precede an obligatory intonational phase break. These experiments showed that (1) there is a smaller word-final effect distinct from the substantial phrase-final lengthening at intonational boundaries, and (2) the word-final effect must also be a boundary effect rather than a result of stress timing, since it occurred even in the *pop opposed–poppa posed* utterances, where the target syllable was always followed by exactly one unstressed syllable. The second set of experiments tests whether word-final lengthening is affected by pitch-accent and stress placement, or whether it defines a phrasal unit independent of metrics.

EE16. Accents, focus distribution, and the perceived distribution of given and new information: An experiment. S. G. Nooteboom (Institute for Perception Research, Eindhoven, The Netherlands and the Department of General Linguistics, Leyden University, Leyden, The Netherlands) and J. G. Kruyt (Institute for Dutch Lexicology, Leyden, The Netherlands)

This paper reports on an experiment examining some perceptual consequences of correspondences between accent patterns, plus and minus focus, and new and given information in Dutch-spoken sentences. The main question examined was to what extent plus focus (accented) constituents are generally perceived as conveying recently introduced ("new") information, and minus focus (unaccented) constituents as conveying earlier introduced ("given") information. The linguistic material for the experiment was formed by brief radio news items, each two sentences long. First sentences determined the distribution of new and given information in second sentences. The accent patterns in the second sentences were varied systematically by manipulating their synthetic pitch contours according to the rules for Dutch intonation. Subjects were asked both to find a fitting first sentence to each second sentence, and to rate the acceptability of each possible combination of a first with a second sentence on a scale from 1–10. Results showed that the most preferred or acceptable distribution of new and given information closely matches the distributions of plus and minus focus, as marked in the accent patterns. New information can hardly ever be acceptably associated with minus focus, but given information can, rather often, but not always, be acceptably associated with plus focus.

EE17. Interacting effects of preceding stress and phonemic vowel quantity on consonant duration in Finnish. Riitta Välimaa-Blum (Department of Linguistics, Ohio State University, 1841 Millikin Road, Columbus, OH 43210)

Finnish has a quantity opposition in both vowels and consonants, and demarcative stress on the initial syllable. Stressed vowels are longer than corresponding unstressed ones. A peculiar fact about short and long vowels is that they are significantly longer after a short initial syllable [CV.CV(V)] than after any kind of long initial syllable [CVV.CV(V), CVC.CV(V), CVVC.CV(V)]. A proposed explanation is that duration as a stress cue spills over to the second vowel in CV.CV(V) words in order to avoid neutralizing the phonemic vowel quantity on the first syllable. If this indeed is the source of the second-vowel lengthening, then we would expect a similar effect in syllable final consonants: The consonant after a short stressed vowel (CVC.) should be longer than the consonant after a long stressed vowel (CVVC.). The duration of geminates was studied in four word structures, CVC.CV(V) and CVVC.CV(V), and they were found to be longer in the former group than in the latter.

EE18. Metrical structure of initial syllables in English. Anne Cutler (MRC Applied Psychology Unit, 15 Chaucer Road, Cambridge CB2 2EF, United Kingdom) and David Carter (Computer Laboratory, University of Cambridge, Corn Exchange Street, Cambridge CB2 3QG, United Kingdom)

Cutler and Norris [J. Acoust. Soc. Am. Suppl. 1 **77**, S39 (1985)] provided evidence for a segmentation strategy in continuous speech perception, whereby a word boundary is postulated, and a lexical access procedure initiated, at each metrically strong syllable. In the present study, the likely success of this strategy was estimated against the characteristics of the English vocabulary. Two computerized dictionaries and two frequency counts were consulted. The dictionaries list approximately three times as many words beginning with strong syllables (i.e., syllables containing a full vowel) as beginning with weak syllables (i.e., syllables containing a reduced vowel). Frequency counts reveal that words beginning with strong syllables have a mean occurrence frequency nearly twice that of words beginning with weak syllables. These findings motivate an estimate for everyday speech recognition that approximately 85% of lexical words (i.e., excluding function words) will have strong initial syllables. This estimate was tested against a corpus of 190 000 words of spontaneous British English conversation [J. Svartvik and R. Quirk, *A Corpus of English Conversation* (Lund, Gleerup, 1980)]. Ninety percent of lexical words in this corpus were found to begin with strong syllables. [Research supported by the Alvey Directorate, United Kingdom.]

EE19. On the resyllabification of /l/ in English. Richard Sproat and Toni Borowsky (AT&T Bell Laboratories, 2D-443, 600 Mountain Avenue, Murray Hill, NJ 07974)

English /l/ is known to be light ([l]) or dark ([ɫ]) when syllable initial or syllable final, respectively. The acoustic difference is that [l]s have lower $F1$s and higher $F2$s than [ɫ]s. The effects of varying strengths of boundaries on the resyllabification of /l/ in English have been examined. Speakers were asked to produce two sets of sentences, both of which included /l/s in the vowel contexts /i_ɪ/, /u_ə/, and /o_ə/: (1) The control set had /l/s that were either necessarily syllable initial or syllable final; and (2) the test set had morpheme final /l/s with a vowel initial morpheme following and boundaries of varying strengths [+, #, ## (compound boundary), major phrase boundary, major intonational boundary] intervening. The data were analyzed by two techniques, namely traditional spectrograms, and by measuring the frequencies of the LPC-poles for the /l/s. It was found, for at least some speakers, that light /l/s are possible before all boundaries except a major intonational boundary. These results support the position that resyllabification is generally allowed across any boundary except a major intonational one. We have no data confirming the claim of Halle and Mohanan ["Segmental phonology of Modern English," *Linguistic Inquiry* **16**, 57–116 (1985)] that speakers resyllabify across compound boundaries but fail to do so across word or phrase boundaries.

EE20. Acoustic study of Russian vowel allophones. Patricia A. Keating (Linguistics Department, UCLA, Los Angeles, CA 90024)

The five Russian vowels vary in auditory quality depending on their consonant contexts; one traditional reference gives several major and minor impressionistic allophones for each stressed vowel. The consonants are reported to be relatively unaffected by their vowel contexts, and to block vowel-to-vowel interactions. This paper presents a study of Russian vowels in CV, VC, and CVC syllables for three speakers. Preliminary results for vowels /i/ and /a/ across a wide variety of contexts suggest that the allophonic variation is of three kinds. First, there is target variation: Each of these vowels has one target in some contexts, and a second target in other contexts. Second, some additional vowel variation is due to undershoot of those targets after extensive $F2$ and $F3$ transitions. Finally, however, in many contexts the formants do reach their targets, despite the

THURSDAY AM

extensive transitions. In these cases, the perceived vowel quality variations seem to arise from differences in the transitions alone. [Work supported by NSF.]

EE21. Coarticulation in VCV sequences in Arabic. Lutfi Hussein (Department of Linguistics, Ohio State University, Columbus, OH 43210)

Vowel-to-vowel coarticulation in VCV utterances has been the subject of several studies. Ohman [J. Acoust. Soc. Am. **39**, 151–168 (1966)] found that vowels in VCV utterances in English and Swedish have some transconsonantal effect on one another. He also found some evidence suggesting that secondary articulation features, like palatalization in Russian, block coarticulation. Action theorists, such as Fowler [J. Exp. Phon. Gen. **112**, 386–412 (1983)], explain V-to-V coarticulation in terms of timing; that is, they claim that vowels in speech production are underlyingly overlapping and consonants ride on top of the vowels. This suggestion implies that intervocalic consonants, whether they have secondary features or not, do not block coarticulation. Keating [UCLA WPP **62**, 1–13 (1985)], on the other hand, explains it in terms of autosegmental phonology. She places the features for vowels and consonants on two separate tiers, and leaves consonants unspecified for vowel features, so that V-to-V coarticulation affects an interpolation between vowel targets. Keating's model implies that consonants that have secondary articulation (i.e., vowel features) must block coarticulation. A preliminary analysis of VCV utterances in Arabic shows that pharyngealized consonants indeed tend to block coarticulation, a finding which supports Keating's prediction over Fowler's.

EE22. An acoustic study of the semivowels /w,y,r,l/ in American English. Carol Espy-Wilson (Room 36-545, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

The semivowels have acoustic properties that are similar to both vowels and other consonants. Ways to quantify acoustic cues that distinguish between the semivowels and separate the semivowels as a class from vowels and other consonants have been studied. To make these distinctions, the formant frequencies within the semivowels, the formant transitions between the semivowels and adjacent vowels, and relative energy measures felt to be correlates of such features as voiced, sonorant, nonsyllabic, and stoplike have been looked at. Some results of this study agree with the findings of previous studies. Namely, given minimal pair words, $F1$ separates the glides /w/ and /y/ from the liquids /l/ and /r/, $F2$ separates /w/ from /r/ and /l/ from /y/, and $F3$ separates /r/ and /l/. In addition, it was noted that the cues usually separate the semivowels from vowels and other consonants. In most cases where semivowels cannot be clearly distinguished on the basis of one or more features, considerable acoustic evidence of feature assimilation between them and the adjacent sound(s) exists. For example, often in words like "cartwheel" and "Norwegian," where a vowel is followed by a postvocalic, but not word-final /r/, either the lowest point of $F3$ occurred at the beginning of the vowel or $F3$ remained at the same low frequency throughout the vowel and /r/ segments. That is, the features of the vowel and following /r/ overlapped. Results are based on measurements of 233 polysyllabic words that contain the semivowels in a variety of phonetic environments. Each word was spoken by two males and two females. [Work supported by a Xerox Fellowship.]

EE23. PHONASCII: An ASCII-based system for detailed phonetic transcription. George D. Allen (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

An ASCII-based system is required in order to store phonetic transcriptions in computer memories. Although several phoneme-level systems (e.g., ARPABET) already exist, none of these permits the level of detail required for close transcription. Under the partial sponsorship of CHILDES (Child Language Data Exchange System, Carnegie–Mellon University), an IPA-to-ASCII system has been developed to permit the archiving of highly detailed transcriptions. This system will be described, and its relationship to other, similar projects (e.g., Alvey) will be discussed.

EE24. Glottal source–vocal tract acoustic interaction. Gunnar Fant and Qi-guang Lin (Department of Speech Communication and Music Acoustics, Royal Institute of Technology (KTH), S-100, 44, Stockholm, Sweden)

Earlier work reported in the STL–QPSR (Speech Transmission Laboratory, Quarterly Progress and Status Report, KTH, Stockholm) each year since 1979 has been extended to the modeling of the supraglottal impedance to include five formants and a time span of analysis to embrace several successive glottal periods. The acoustic interaction is highly nonlinear and may give rise to spectral peaks between formants. This is largely due to nonlinear superposition ripple originating from previous excitations. At a high $F0$, coinciding with $F1$, the air consumption is minimized as previously shown experimentally by Rothenberg. One aspect of the interaction is intraglottal variations in formant bandwidths and frequencies. The modeling of superposition at increasing $F0$ and constant articulation shows how the formant amplitude range at in-phase and out-of-phase superposition decreases due to intraglottal losses. In addition, there appear nonlinear deviations such as a weak tendency of synchrony between the second and first formant amplitude dependency of $F0$. A constant leak added to the normal glottal area function has been modeled as a first step toward the study of leaky voices. Here, the apparent linear-type formant ripple in the maximally closed phase is a memento for inverse-filtering criteria. Implications for female and child phonations are discussed.

EE25. Possible production differences in children's and adults' fricatives. Richard S. McGowan (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511) and Susan Nittrouer (Boys Town National Institute, 555 N. 30th St., Omaha, NE 68131)

Most developmental studies of articulatory ability and phonological organization use acoustic measurements to make inferences about production differences between children and adults. A common assumption of this approach is that the only physical differences between children and adults are the lengths of various vocal tract regions. Accordingly, any acoustic difference not accounted for by this scaling is attributed either to a lack of articulatory skill or to poorly defined phonological categories. This report considers whether other physical differences might account for some of the acoustic differences between children and adults. Speech samples of 12 speakers (8 children and 4 adults) producing the fricatives /s/ and /ʃ/ followed by the vowels /i/ and /u/ were analyzed to locate the major spectral prominences. Acoustic measurements showed that the fricative low-frequency prominences for children's samples differed from those of adults in three important ways. They: (1) were generally higher in frequency; (2) were relatively greater in amplitude; and (3) showed greater effects of vowel context. The first finding can be explained by a simple scaling of adult models of fricative production to accommodate children's smaller vocal tracts. The last two findings suggest, however, that there are other anatomical and articulatory differences between children and adults affecting fricative production. The data suggest the possibility that one important difference may be that the ratio of the glottal area to fricative constriction area tends to be less for children than for adults. [Work supported by NIH Grants NS-07237 and NS-13617 to Haskins Laboratories.]

EE26. Sex differentiation cues in the voices of young children of different language background. Inger Karlsson (Department of Speech Communication and Music Acoustics, Box 700 14, KTH, S-100, 44, Stockholm, Sweden) and Martin Rothenberg (Department of Electrical and Computer Engineering, Link Hall 111, Syracuse University, Syracuse, NY 13210)

Children, aged 3–8 years, speaking different native languages, were recorded and speech samples from these children were played back to adult listener groups, also of different language background. The listeners were asked to judge the sex of the child. The children were native speakers of English, Finnish, or Swedish and the listeners were native speakers of Finnish, Swedish, English, or Chinese. Speakers and listeners from different language backgrounds were used to find out if the differences in boys' and girls' voices were socially acquired or innate. The average right judgment was 66% over all speakers and listeners, with no strong tendency for a higher percentage of right answers for children speaking the listener's own language. The results from the listening tests have been compared to different acoustic measurements of $F0$, vowel formants, and speaking rate. No apparent correlation between perceived sex and these measurements has been found.

EE27. A multidimensional scaling study of normal voice quality in females. Marylou Pausewang Gelfer (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Little is currently known about the perceptual dimensions listeners use in discriminating female voices and how these dimensions relate to perceptual rating-scale judgements. The purpose of this study was to examine these issues for normal female voices. In addition, the judgments of 20 naive listeners and 20 experienced listeners (speech pathologists) were compared. Listeners provided dissimilarity judgments of sentences produced by 20 females (24–25 years). The listeners also rated each voice on 17 perceptual rating scales (e.g., high pitch–low pitch, clear–hoarse). Results of the multidimensional scaling (MDS) analysis produced a five-dimensional solution (RSQ = 0.509). A stepwise linear regression related the perceptual attributes of pitch, age, loudness, "liveliness," and pitch variability/quality to the five MDS dimensions. When listener groups were compared, the experienced listeners relied on pitch somewhat more than the naive listeners, while the naive listeners utilized the pitch variability/quality dimension to a greater degree. This study demonstrates that listeners' perceptual ratings can be significantly correlated with perceptual dimensions derived from MDS analysis. [Research supported by BRSG Grant S07 RR07031, DRR, NIH.]

EE28. Age–sex variations in maximum performance measures of speech production. Ray D. Kent and Jane F. Kent (Department of Communicative Disorders, University of Wisconsin, Madison, WI 53706)

The maximum performance measures of speech production include: phonation volume, maximum phonation time, maximum expiratory pressure, maximum vocal SPL, physiological vocal frequency range, maximum articulatory force, and maximum syllable repetition rate. A survey of the normative data seemed in order, given the frequent clinical use of these measures and the availability of a large, although scattered, archival database for normal speakers. This report reviews published data for each measure and considers the adequacy of these data for clinical application. The data are found to be deficient in several respects: (1) the effects of subject age and sex are incompletely documented; (2) test procedures are not standardized; (3) the validity of a maximum performance estimate is open to question for most, if not all, measures; (4) the variability of the data within and across studies is very large; and (5) a theoretical framework for the interpretation of the measures is either lacking or deficient. Recommendations are offered to correct some of these problems, but the major conclusion of this report is that measures of maximum performance lack validity and efficiency for general clinical application. Summary tables of the normative data are available upon request. [Work supported by NIH.]

EE29. Variability differences in the speech production in two age groups of adult females. Richard J. Morris (Department of Communication Disorders, University of Utah, Salt Lake City, UT 84112) and W. S. Brown, Jr. (IASCP-ASB 63, University of Florida, Gainesville, FL 32611)

The purpose of this study was to examine the variability between the speech produced by two age groups of adult females, a young adult group and a geriatric group. The 25 women in each group phonated three productions of the vowel /a/ for 5 s at three effort levels—maximum, comfortable, and minimum. Recordings of these productions were digitized and analyzed for mean intensity and intensity distribution. The women also repeated a series of CV, VCV, and VC syllables using the consonants /p/, /b/, /t/, /d/, /s/, and /z/ combined with the vowel /a/, in the carrier phrase "Speak ____ again." Intraoral air pressure and the voicing signal were recorded to provide vowel and consonant durations, voice onset times, and peak intraoral air pressure. The F-max test for homogeniety of variance was used to compare the variability between the two age groups. The older speakers exhibited significantly greater variability than the younger speakers for the dependent variables, peak intraoral air pressure, consonant duration, vowel duration, voice onset time, and the minimum effort level of speech intensity. The greater variability of production exhibited by the older speakers could be a factor in the perception of older speakers.

EE30. Acoustic–phonetic correlates of language development. Harold R. Bauer and Lauren Nelson (Speech and Hearing Science Section, Department of Communication, Ohio State University, 324 Derby Hall, 154 North Oval Mall, Columbus, OH 43210)

Phonetic contrast has been suggested as a cognitive operating principle underlying the continuity of infant phonetic development [H. R. Bauer, J. Acoust. Soc. Am. Suppl. 1 **75**, S45 (1984)]. Phonetic contrast was estimated in the current study using the phonetic product (PP) estimator weighted to the same eight American English sound-class frequencies. The PP was a result of the multiplication of the number of eight different contrasting sounds per minute after weighting and was summarized from recordings of mother–infant interactions. The results from four children were compared when each was 1 and 2 years old. Associated measures of language development at 2 years of age were also compared for predicted order based upon the subject's 1-year data. As an estimator of phonetic contrast, PP and its acoustic correlates in 1-year-old infants, were predictive of most of their PP and associated language variables at 2 years of age.

EE31. Vowel context effects in children's and adults' speech. Joyce Powers (Department of Linguistics, Ohio State University, Columbus, OH 43210)

Recent studies comparing coarticulation in adults' and children's speech have yielded contradictory results; some have indicated that children coarticulate more, and others that they coarticulate less. The studies which found more coarticulation in children's speech involved fricatives and the vowels /i/ and /u/; those finding more coarticulation among adults involved velar stops and the vowels /i/ and /a/. It may be that coarticulation is not a unified phenomenon, and that the conflicting results of earlier studies were a consequence of studying different kinds of coarticulation. The present study, therefore, considers all of the things that were separately investigated before. Adults and 3- and 4-year olds were recorded producing the syllables /si/, /su/, /sa/, /ʃi/, /ʃu/, /ʃa/, /ki/, /ku/, and /ka/. The spectral peak of the fricative or stop will be determined, as will the F_2 values at the onset of voicing of the vowel. The results of the study will be considered in terms of what they imply about how children acquire the phonological units of their language.

THURSDAY AM

EE32. Cross-language study of children's prosodic development—IV: Comparison of 4- and 5-year-old French-, German-, and Swedish-speaking children's TAKI productions. Denise Luczycki and George D. Allen (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

Six 4-year-old and six 5-year-old French-, (Austrian) German-, and Swedish-speaking children were engaged in an experimental task requiring them to first learn and then produce the names of toys. Each pair of toys was associated with a pair of names, which differed in prosodic contour, one pair being phonologically simple (/'taki/ versus /ta'ki/), one slightly more complex (/'smopfi/ versus /smop'fi/), the third much more complex (/'gluvistrap/ versus /gluvi'strap/). The children's spontaneous productions of these names were tape recorded and analyzed auditorily and spectrographically for their prosodic features ($F0$, duration, and intensity). Productions of the prosodically simpler forms were

more likely to be prosodically correct than the more complex forms, for all children. The pattern of errors was different, however, among the three language groups. For example, the French children tended to drop final weak syllables, whereas the Austrian children showed a reverse tendency, dropping initial weak syllables, and the Swedish children did neither. Each of these error patterns conforms to the phonology of the children's target language. [Work supported in part by NSF.]

EE33. Anticipatory coarticulation in children with good and poor syntactic abilities. Philip Lieberman and Joan A. Sereno (Department of Cognitive and Linguistic Sciences, Box 1978, Brown University, Providence, RI 02912)

The present study investigates the relation between speech coarticulation and syntactic processing in children. Specifically, acoustic and perceptual measurements of anticipatory labial and lingual coarticulation are compared to the syntactic abilities of children, as measured by the Rhode Island Test of Language Structure. Two children with good (greater than 95% correct) and two children with poor (less than 85% correct) syntactic comprehension scores produced five tokens each of the following CV syllables ([si], [su], [zi], [zu], [ti], [tu], [di], [du], and [ki], and [ka]). The effects of the vowel on the preceding consonant were acoustically assessed by measuring both second formant and characteristic spectral prominence regions in the consonantal stimuli. Perceptual measurements were then performed by presenting these excised consonantal segments for vowel identification. The correlation between the coarticulation data and the measurements of the children's syntactic abilities is evaluated. The results are discussed in terms of theories of motor programming, the evolution of the hominid brain, and the relationship of speech motor programming to the development of syntax.

EE34. The effect of speaker gender on word recognition. Jan M. Wasowicz and Lois L. Elliott (Department of Communication Sciences and Disorders, Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

The effect of speaker gender on word recognition was investigated using a forward-gating paradigm [F. Grosjean, Percept. Psychophys. **28**, 267–283 (1980)]. Stimuli varied in number of syllables (1 or 2) and gender specificity (male-, female-, or neutral-oriented). The stimuli constituted the final word of a declarative sentence (e.g., "I recently bought a new pipe/dress/bike"). Each sentence was spoken by both a male and a female talker, recorded and digitized before the final words were gated. The shortest gate for each word was the initial 30 ms. Successive gates were increased in duration by 30 ms. The purpose was to determine whether words would be identified at shorter gate durations when the word was appropriate to the speaker's gender, as compared to when it was not. The results suggested that speaker gender facilitated the recognition of gender-appropriate words and inhibited the recognition of gender-inappropriate words. Results will be discussed with respect to current models of word recognition. [Work supported by Dissertation Year Grant, Northwestern University.]

THURSDAY MORNING, 14 MAY 1987 — CELEBRATION HALL, 9:00 TO 11:30 A.M.

Session FF. Bioresponse to Vibration II and Physical Acoustics V: Biophysical Acoustics: Absorption in Biological Media I

Frederick Kremkau, Chairman
Bowman Gray School of Medicine, Center for Medical Ultrasound, 300 S. Hawthorne Road, Winston-Salem, North Carolina 27103

Invited Papers

9:00

FF1. Theoretical models of ultrasound absorption in tissues. James F. Greenleaf and Chandra M. Sehgal (Biodynamics Research Unit, Mayo Clinic/Foundation, Rochester, MN 55905)

On the basis of many measurements on attenuation, absorption, and speed of ultrasound in biological materials, one can make the following generalizations: (a) Biological tissues have high attenuation coefficients, about two to three orders greater than their major component, water; (b) the amplitude attenuation coefficient increases with frequency near-linearly in soft tissues and near-quadratically in fluids with frequency in the range 1 to 10 MHz, and possibly even up to 100 MHz; (c) speed dispersion is extremely small ($\sim$0.1 to 1 m/s/MHz); (d) absorption and attenuation coefficients either increase or decrease with an increase in temperature, depending on the nature of the tissue, and the frequency of irradiation; (e) fluids containing proteins show a sharp increase in excess absorption in acidic and basic *p*H ranges, and finally; (f) at sufficiently high intensity of irradiation the absorption coefficient increases with intensity. Comprehensive models for the propagation of ultrasonic waves through lossy media should explain these experimental findings. We will review some of the current models and critically examine them in light of the definition of absorption in relation to attenuation and scattering. [Work supported by NIH CA 24085, CA 41324, and NSF 8310626.]

9:30

FF2. Biomolecular ultrasound absorption. F. Dunn (Bioacoustics Research Laboratory, University of Illinois, 1406 W. Green Street, Urbana, IL 61801)

Studies of the bulk ultrasonic properties of intact tissues reveal the importance of biomacromolecules in determination of ultrasonic absorption. The absorption coefficient of blood has been found to be proportional to the weight percentage of the protein content, which has encouraged studies of the absorption by proteins and their analogs in solution. Studies of amino acids, polyamino acids, and soluble proteins in aqueous solution have suggested such mechanisms as relaxation of proton-transfer equilibria occurring at ionizable residues and relaxation of solvent–solute equilibria. Though other mechanisms have been proposed, no single mechanism appears to be predominant in being responsible for ultrasonic absorption by proteins.

10:00

FF3. The contribution of absorption to tissue attenuation. Kevin J. Parker (Department of Electrical Engineering, University of Rochester, Rochester, NY 14627)

The relative contributions of ultrasonic scattering and absorption to the overall attenuation coefficient of tissues is an important issue in diagnostic and therapeutic ultrasound. This paper presents a range of experimental evidence and case studies which support the postulate that absorption dominates attenuation in soft tissues. Comparisons of tissue absorption coefficients (using thermal pulse decay) and attenuation coefficients (using radiation force insertion loss) are given, along with results of tissue homogenization experiments. Other evidence is gained from measurements of total scattered power, and clinical observations of backscatter and attenuation of focal lesions. Finally, comparisons are made of the attenuation of dilute protein solutions and whole liver specimens, in order to establish the role of multiple relaxation mechanisms at the macromolecular level in the overall attenuation coefficient of mammalian liver tissue.

10:30

FF4. Calculation of temperature elevation caused by ultrasound absorption. Wesley L. Nyborg (Physics Department, University of Vermont, Burlington, VT 05405)

It has been known for decades that ultrasound is capable of producing physiological change in man and in laboratory animals. A dominant mechanism for the change is temperature elevation resulting from sound absorption. It has been shown for some situations that the processes for heat generation and transport are understood well enough to make it feasible to predict the temperature that results from specific acoustic exposures. Thus Pond, Robinson and Lele, Carstensen, Lizzi, and others have successfully calculated intensity–time combinations required for production of recognizable lesions in mammalian tissues and other media. There is now special interest in computational methods for application to ultrasonic hyperthermia, and also to the formulation of safety criteria for diagnostic ultrasound. For the latter purpose a simplified method has been developed, based on a Green's function solution to the bio-heat transfer equation. In this paper, previous work on temperature calculations is reviewed, and recent results discussed.

11:00

FF5. Energy transmission into the human hand from vibrating tools. Douglas D. Reynolds (Department of Civil and Mechanical Engineering, University of Nevada–Las Vegas, Las Vegas, NV 89154)

A method for calculating power transmitted to the hands of operators who use vibrating hand tools is presented. Results that relate to a comprehensive multidisciplined NIOSH field study of several hundred chipper and grinder workers who used pneumatic hand tools are discussed. These results indicated that the total power in the frequency range of 6.3 to 1000 Hz transmitted to the hand ranged from 1080 to 7230 J/s for the chisel and from 0.852 to 157 J/s for the handle of chipping hammers. For pneumatic grinders the power transmitted to the hands of the tool operators was in the range of 0.00658 to 0.235 J/s over the same frequency range.

Session GG. Shock and Vibration III: Machinery Foundations and the Transmission, Isolation, and Damping of Structural Vibrations

Louis A. Herstein, Chairman
Naval Sea Systems Command, Ship Silencing Division: Code 55N, Washington, DC 20362

Chairman's Introduction—9:00

Invited Papers

9:05

GG1. A design optimization for shipboard machinery foundations subjected to noise and shock requirements. Russel Miller, Ken Nuss, Peter Kasper, and George Amir (NKF Engineering, Inc., 12200 Sunrise Valley Drive, Reston, VA 22091)

The overall methodology used to design foundations are sometimes a compromise between the various conflicting requirements of noise, shock, vibration, ship motion, thermal motion, and machinery imposed loads. The methodology developed is based on iterative structural design to achieve optimum stiffness and the predominant natural frequency that best mitigate shock and vibration motions prescribed by realistic shipboard environment. Experimental data of tested foundations in the prescribed shock environment is used to verify the validity of the structural optimization method developed. The paper describes the computer code developed for the structural optimization. Selection of various foundation categories are available for various locations in the ship and various installation configurations. Comparisons between structurally optimized and nonoptimized foundations are made demonstrating substantial cost and weight saving as well as improved hardness and vibration reduction. The iterative process of structural optimization for shock and vibration is linked to noise attenuation performance. The analysis of noise transmission through the foundation was performed using the direct dynamic stiffness method (DDSM). This approach provides velocity, force, and power transfer functions between input excitation positions and hull intersection positions.

9:35

GG2. Effectiveness of structureborne noise isolation systems on a lightweight ship foundation. G. L. Fox (Fox Engineering Analysis, 4455 Torrance Boulevard, Suite 196, Torrance, CA 90503) and Terry D. Scharton (ANCO Engineering, 9937 Jefferson Boulevard, Culver City, CA 90230)

Radiated structureborne noise from main propulsion machinery in a lightweight, high-performance craft is of concern. This is due to the conflicting requirements of human habitability and ship weight. Efficient structureborne noise isolation of the main machinery will reduce the weight of the acoustical treatment required in receiver spaces. Main machinery isolation involves the consideration of the mechanical vibratory interactions between the machine, subbase, and foundation as well as the elastic mountings that connect these components. This paper presents a method that allows an analyst to calculate the structureborne noise power transmitted into a structure composed of idealized finite length beams. Impedance relationships are developed for the machine, subbase, foundation, ship hull, and the intervening isolators. Idealized finite length beam impedances are derived for various boundary conditions. The results of a parametric study to assess the effectiveness of vibration isolators, for various system parameters, is presented for a ship hull of typical aluminum construction.

10:05

GG3. Selection of resilient mounts for protection of fragile equipment considering foundation and equipment structural admittances. Harry Himelblau (M.S. 301-456, Jet Propulsion Laboratory, Passadena, CA 91109) and Sheldon Rubin (M.S. M4/899, The Aerospace Corporation, P.O. Box 92957, Los Angeles, CA 90009)

In 1962, a group of structural dynamicists under the auspices of SAE Committee G-5 on Aerospace Shock and Vibration developed a procedure on the above subject, which was published as SAE Doc. AE-3. This procedure was restricted to the assumption of rigid equipment structure under translational excitation and response. Five years later in an unpublished report, Rubin extended the procedure to permit the consideration of arbitrary admittances of both equipment and foundation. This extension is reviewed in this presentation, including a simple criterion for mount adequacy, a graphical method for verifying that adequacy, and an overlay for the rapid selection of mount stiffness and damping. Mount restrictions to avoid bottoming under sinusoidal and random excitation are also discussed.

10:35

GG4. Vibration damping performance—What we should know about it. Pranab Saha (Kolano and Saha Engineers, Inc., 1899 Orchard Lake Road, Suite 105, Pontiac, MI 48053)

The vibration damping performance of materials that are used in the automotive industry are usually tested by either: (1) the Geiger Plate Test Method, or (2) the Complex Modulus Test Method. The first test method has some advantages of testing materials that are unique to the automotive industry. Both of these methods have certain limitations, though the second method is a superior one in the sense that it can measure the damping performance with temperature and frequency. This paper discusses the limitations of some of the ways the data are typically presented using this method. The paper also discusses how design charts can be used to overcome these limitations and compute the damping performance of the composite material (composite loss factor) for a given application.

10:50

GG5. Acoustic characterization of some commercially available closed cell rubbers and corprene. Gary Caille, Jacek Jarzynski, and Peter Rogers (The George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Three closed cell and two corprene rubbers (manufactured by Rubatex and Groendyk Corporations, respectively) with Neoprene and nitrile bases have been experimentally investigated in the frequency range of 10 to 70 kHz. These materials contain approximately 50% air and simulate a pressure release surface with reflection coefficients near -0.95. The characterization includes longitudinal sound speed and transmission, reflection, and absorption coefficients as functions of incident angles and frequency. The dynamic (complex) Young's modulus and loss factors have been determined by the longitudinal resonance method. Poisson's ratio is assumed to be 0.499 which allows prediction of the shear modulus. A comparison between the experimentally determined reflection and transmission coefficients with theoretical values [D. L. Folds and C. D. Loggins, J. Acoust. Soc. Am. **62**, 1102–1109 (1977)] is presented. Prediction of the moduli using contemporary composite models will also be presented.

11:05

GG6. Wavenumber/frequency admittance of beams. Karl Grosh, W. Jack Hughes, and Courtney B. Burroughs (Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804)

Using an array of point drives, the admittance of finite beams were measured and compared to predictions. The predictions were made using an analytic model of the beam response based on Timoshenko beam theory. To drive the desired wavenumbers, the amplitudes of the point drives were weighted to produce standing wave patterns that were made up of left and right traveling waves, both of the same wavenumber. The standing wavepattern avoids the need to control the phases between the drives that would have been required for traveling single wavenumber drives. The effects of beam resonances are explored by comparing measured admittance data for beams with free ends, with and without internal damping and for beams with heavily-damped ends. The wavenumber/frequency transfer functions of resilient layers placed between the array of drives and the beam are determined from admittance measurements made with and without the resilient layer.

11:20

GG7. Dynamics of piezoelectric rods under initial stresses. M. Cengiz Dökmeci and E. E. nanI'D (Istanbul Technical University, P.K.9, Istanbul 80191, Turkey)

The analysis presented is an extension of our earlier treatment of rods [Int. J. Solids Struct. **8**, 1205–1222 (1972); and **10**, 401–409 (1974)] in which the effect of initial stresses is neglected. In this study, this effect is considered on waves and vibrations of piezoelectric thin rods. First, a variational principle due to the author [Proc. 39th Freq. Control Symp., pp. 431–435 (1985)] is reformulated, and the field quantities are expanded in terms of Legendre polynomials. Then, the expansions together with the variational principle are used so as to consistently establish a hierarchical system of one-dimensional equations of crystal rods. The governing equations may incorporate the higher-order effects of piezoelectric rods by a proper truncation of expansions. They accommodate all the types of extensional, flexural, and torsional motions of rods. Further, the resulting equations are considered for special motions, and they are shown to agree with those of earlier ones. [Work supported in part by the U.S. Army through its European Research Office.]

11:35

GG8. A study of a parametrical system with a clearance type nonlinearity. R. Comparin and R. Singh (The Ohio State University, Department of Mechanical Engineering, 206 West 18th Avenue, Columbus, OH 43210-1107)

Gear rattle is one of the major noise and vibration problems affecting lightly loaded drives. Gear rattle is a complicated, nonlinear vibration, characterized by backlash induced vibro-impact between meshing gears. For geared systems with a constant driving torque, rattle is the result of a parametric excitation due to mesh stiffness variation and transmission error. This work represents a fundamental study of the vibro-impact problem in a parametrically excited mechanical system with clearance. Specifically, this work describes the frequency domain characteristics of a parametrically excited single degree of freedom system with a clearance type nonlinearity. The stability of steady-state solutions are also considered. An approximate analytical technique is used and results are compared with digital (numerical integration) simulation results.

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

Technical Advisory Group for ISO/TC 43/SC1 Noise

K. M. Eldred, Chairman S12
P.O. Box 1037, Concord, Massachusetts 01742

H. E. von Gierke, Chairman, Technical Advisory Group for ISO/TC 43/SC1
Biodynamics & Bioengineering Division, AFAMRL/BB U.S. Air Force, Wright-Patterson AFB, Dayton, Ohio 45433

Standards Committee S12 on Noise. Working group chairs will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC1 activities will be discussed, with a report on the meeting of ISO/TC 43/SC1 which was held in Copenhagen, Denmark from 4–7 May 1987.

Session HH. Psychological and Physiological Acoustics VIII: Ear Canal Measures and Hearing Impairment

Honor O'Malley, Chairman
Audiology Program, Teachers College, Columbia University, New York, New York 10027

Contributed Papers

1:00

HH1. Auditory system impedance measurement using the two-microphone wave-tube technique. Douglas H. Keefe (School of Music DN-10, University of Washington, Seattle, WA 98195), Edward M. Burns, and Robert Ling (Department of Speech and Hearing Science JG-15, University of Washington, Seattle, WA 98195)

An impedance measurement technique suitable for middle ear investigations has been devised that has the advantages of fast data acquisition (less than 1 s) and large measurement bandwidth (200–6000 Hz). This technique does not require a transducer delivering a constant volume flow. The excitation used is a broadband chirp whose spectral shape may be varied to enhance signal to noise ratio. The method depends upon pressure measurements at two points within a smooth, cylindrical tube whose transmission characteristics are well known. In our implementation of this technique, the frequency-dependent viscous and thermal losses in the wave tube are accounted for. The wave tube is inserted into the ear canal, and the input impedance at the tube tip is computed based upon the tube transmission coefficients and the measured transfer function between the microphones. Measurements are presented of the input impedance in normal adult ears for comparison with the literature. Since the technique is fast and relatively noninvasive, it appears well-suited for our project goal of measuring the middle ear impedance of infants. [Work supported by NINCDS.]

1:15

HH2. Two-microphone measurement of acoustic intensity in the ear canal as a calibration of high-frequency hearing. William J. Murphy, Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907), and Glenis R. Long (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

As an alternative to measures of ear canal pressure in the vicinity of the tympanic membrane, we propose an acoustic intensity measurement slightly downstream from the point of sound delivery as a convenient and meaningful calibration of high-frequency hearing. If sound absorption in the air medium and at the walls of the ear canal is neglected, the power flow through any cross section of the ear canal would be equal to the power flow into the middle ear. A sound delivery and intensity measurement system has been built using a two-port earmold. The tube of an Etymotic Research ER-2 sound delivery system passes through one port and a pair of probe tubes (outer diameters = 1 mm, tip separation ≈ 3 mm) pass through the other. The latter are connected to ER-7 microphones and used for acoustic intensity measurements based on a cross-spectrum algorithm. Data are given on the sensitivity of measurement results with respect to the relative positions of the sound delivery point and probe tube tips. [Work supported by a grant from the Deafness Research Foundation.]

1:30

HH3. High-frequency audiometry and intersubject variations of ear canal geometry. Michael R. Stinson (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

An extension of current audiometric techniques to frequencies greater than 8 kHz is highly desirable but difficult because of the large intersubject scatter in hearing threshold data at these frequencies. One of the main sources of scatter is the variation between subjects of size and shape of ear canal. This factor is examined by considering a model of an audiometric headphone mounted over the ear of a subject. The ear canal is treated in detail using a horn equation approach to calculate the pressure distribution along its length, given its exact geometry; the calculation can be repeated for different canal geometries with everything else (headphone, eardrum impedance) unchanged. Above 8 kHz the transformation from headphone input voltage to sound pressure at the eardrum is a function which depends on frequency, showing peaks at frequencies that depend critically on the choice of geometry. For a realistic range of ear canal lengths and shapes the eardrum pressure (and, hence, the apparent threshold) can vary by as much as 20 dB between subjects at a specific frequency, for the same input voltage. Conventional audiometric techniques must be modified considerably if they are to be applied above 8 kHz.

1:45

HH4. Diffuse field response of the ear. Mead C. Killion (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007), Elliott H. Berger, and Richard A. Nuss (E-A-R Division, 7911 Zionsville Road, P.O. Box 68898, Indianapolis, IN 46268-0898)

The ear canal pressure developed in a diffuse-field facility complying with ANSI S12.6-1984 was measured on 20 adult subjects (11 male and 9 female), with two replications on each ear for 19 of the 20 subjects. A soft 1-mm-o.d. silicone rubber probe tube was inserted until the subject reported the "thump" of the tube bumping the eardrum, and then withdrawn a couple of mm. Preliminary experiments indicated that this technique did not always produce the same results, with some curves on the same ear of the same subject showing substantially less "eardrum SPL" in the 8- to 20-kHz region than others. It was sometimes (but not always) possible to confirm with otoscopic examination that the lower-SPL curves corresponded to probe-tube locations away from the eardrum. Based on the preliminary experiments, data from the "best run of the best ear" (i.e., highest level above 8 kHz) were averaged. The resulting curve was similar to the curve obtained on a KEMAR manikin in the same sound field.

2:00

HH5. Reference threshold levels for the ER-3A insert earphone. Laura Ann Wilber (Program in Audiology, Northwestern University, Evanston, IL 60202), Barbara A. Kruger (Kruger Associates, 37 Somerset Drive, Commack, NY 11725), and Mead C. Killion (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007)

Two sets of provisional reference threshold sound pressure levels have been provided by the manufacturer for the calibration of the ER-3A insert earphone. One set is for use with the B&K DB-0138 coupler (an HA-2 2-cc coupler that comes with a 5-mm-diam by 2-mm-long stainless steel coupling nipple preceding the 18 mm of 3-mm-diam earmold-simulation portion specified in S3.7-1973), and the other set is for use with the HA-1 coupler (the standard "In The Ear hearing aid" 2-cc coupler) with a foam eartip sealed directly to the top surface of the coupler. Several recent studies have demonstrated that the ER-3A may be directly substituted,

without recalibrating, for a TDH-39/MX-41AR earphone in some cases, but most available data have not been reduced to a form suitable for establishing a revised estimate of the reference threshold levels. This paper reports the results of such a data analysis.

2:15

HH6. The interaction of impulse and continuous noise: Energy and spectral considerations in the production of hearing loss. Michele Roberto (Department of Bioacoustics, University of Bari, Bari, Italy), William A. Ahroon, Robert I. Davis, and Roger P. Hamernik (Auditory Research Laboratories, SUNY, Plattsburgh, NY 12901)

Realistic industrial noise environments containing impulsive and continuous noise were modeled using a 5-day exposure paradigm that produces an asymptotic threshold shift (ATS). Pre- and postexposure measures of hearing thresholds were obtained on 96 chinchillas using evoked auditory responses (EAR). Six control groups were exposed to octave bands of noise at 0.5, 2.0, and 4.0 kHz at 95, 90, and 86 dB SPL, respectively, or impacts of 113, 119, or 125 dB peak SPL presented once per 1, 4, or 16 s, respectively. Nine interaction groups were exposed to combinations of an impulse and continuous noise. The greatest spectral overlap of energy occurs between the impulse and the 0.5 kHz octave band of noise. Although each of the different impulse noise exposures were balanced to produce an equal energy exposure, an exacerbation of hearing loss was produced in animals exposed to the 119- and 125-dB impacts in combination with the low-frequency (0.5-kHz) continuous noise. This synergistic effect gradually disappears when the spectral overlap between noises is reduced. [Research supported by NIOSH.]

2:30

HH7. Temporary threshold shift after exposure to a narrow-band noise, frequency-modulated tones, continuously variable frequency tones and a pure tone. I. M. Young and L. D. Lowry (Department of Otolaryngology, Jefferson Medical College of Thomas Jefferson University, Philadelphia, PA 19107)

Temporary threshold shift after exposure to four stimulating sounds were measured by automatic audiometry for three subjects with normal hearing. Stimulating sounds were (1) a pure tone, (2) continuously variable frequency tones between 1250 and 1750 Hz, (3) frequency-modulated tones centered at 1500 Hz with a frequency deviation of $\pm$ 250 Hz and a modulation rate of 25 per s, (4) a narrow-band noise with a bandwidth of 1250–1750 Hz. Subjects were exposed to stimulating sound through an earphone at the intensity of 110 dB SPL and duration of 10 min. Temporary threshold shift was measured for frequencies between 1000 and 8000 Hz beginning at 5 s after cessation of the stimulating sound. Results indicated that the greatest shift was observed at a frequency 2000 Hz by a pure tone stimulation and the least shift by a narrow-band noise. Effects of continuously variable frequency tones and frequency-modulated tones were intermediate. These findings were compared and discussed with temporary threshold shift following exposure to white noise at equivalent intensity and duration.

2:45

HH8. Effects of brief, intense tones on auditory temporal acuity. Craig A. Champlin and Lawrence L. Feth (Department of Speech–Language–Hearing: Sciences and Disorders, University of Kansas, Lawrence, KS 66045)

A gap detection threshold (GDT) procedure was used to measure auditory temporal acuity in humans before and after exposure to a brief, intense low-(0.4-kHz) or high-(1.7-kHz) frequency tone. The maximal temporary threshold shift (TTS) produced by each exposure was approximately 10 dB. GDT stimuli were octave-band noises centered at one of three frequencies: the exposure frequency, 1/2 oct above the exposure frequency, or 1 oct above the exposure frequency. GDTs were obtained at 35, 55, and 75 dB SPL at each center frequency. GDT and TTS recovery were monitored for 16 min following an exposure. The results from the high-frequency exposure condition indicate that changes in GDT can be predicted from shifts in absolute threshold. For the low-frequency condition, differences between the GDT and TTS recovery functions and elevated GDTs in the absence of significant TTS require an alternative explanation. [Work supported by NIOSH.]

3:00

HH9. Temporal integration in normal hearing, cochlear impairment, and impairment simulated by masking. Mary Florentine, Hugo Fastl, and Søren Buus[a] (Communication Research Laboratory, 133 FR, Northeastern University, Boston, MA 02115)

To assess temporal integration in normal and impaired listeners, absolute thresholds for tones were measured as a function of duration. Durations ranged from 500 ms down to 15 ms at 0.25 kHz, 8 ms at 1 kHz, and 2 ms at 4 and 14 kHz. An adaptive 2I, 2AFC procedure with feedback was used. On each trial, two 500-ms observation intervals, marked by lights, were presented with an interstimulus interval of 250 ms. The monaural signal was presented in the temporal center of one observation interval. The results for four normal and six impaired listeners show (1) as expected, thresholds for normal listeners decrease about 8 dB per decade of duration, (2) for impaired listeners, the decrease usually is much less (2 to 6 dB) in the range of a hearing loss, but normal at frequencies where thresholds are normal, (3) for listeners with impairments simulated by masking, the slopes are nearly the same as those for normal listeners and steeper than those for the real impairments. These results indicate that the shallow slopes observed for impaired listeners probably are not due to splatter of energy to frequency regions where thresholds are low, but reflect diminished temporal integration, *per se*. [Work supported by NIH-NINCDS RO 1NS 18280.] [a] Also at Department of Electrical and Computer Engineering, Northeastern University.

3:15

HH10. Comparison between loudness functions in noise and in noise-induced hearing loss. Rhona P. Hellman (Auditory Perception Laboratory, Northeastern University, Boston, MA 02115)

Pure-tone loudness functions were generated in a cochlear-impaired population with moderate to severe noise-induced losses. Three procedures were used: magnitude estimation, magnitude production, and cross-modality matching. The slope of the loudness function was determined over the stimulus range where cochlear impairment steepens the loudness function. Both the measured and predicted slopes show that the steepening of the loudness function depends on the severity of the hearing loss; that is, the higher the threshold the steeper the function. While this behavior is also observed in partially masked normal ears, the extent of the agreement between the partially masked and impaired loudness functions is determined by the bandwidth of the masking noise. The impaired loudness functions closely agree with those obtained in wide-band noise, but not with those obtained in narrow-band noise as wide as an octave. This finding implies that the effect of the cochlear impairment on the loudness function resembles that of an external noise with a broad frequency spectrum. [Work supported by the Rehabilitation Research and Development Service of the VA.]

3:30

HH11. Loudness judgments in recruiting ears as influenced by intensity, and interspersed changes in intensity. Maureen A. Korman, Ernest M. Weiler, and Angel Dell'Aira (Communication Disorders, University of Cincinnati, Mail Location No. 379, Cincinnati, OH 45221)

It has been found that loudness judgments adapt to monaural stimulus increments [E. M. Weiler, D. E. Sandman, and L. M. Pederson, Brit. J. Audiol. **15**, 201–204 (1981)]. The present study was undertaken to determine whether the changes in high-frequency recruiting ears measured at a nonrecruiting frequency (1000 Hz), were the same as in normal listeners. Both groups showed a similar decline in judgment to the baseline 660-dB tone. There was a significant interaction between groups, intensity, pulsed

duration of an interspersed 10-dB increment, and successive judgments of loudness. Similar interactions were seen at 20 dB above the baseline intensity.

3:45

HH12. Predicting masked thresholds in impaired listeners from combinations of normal masked thresholds and hearing loss. Walt Jesteadt, Patricia G. Stelmachowicz, and Brian P. Callaghan (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131)

Many studies have simulated sensorineural hearing loss using maskers that produce equivalent threshold elevations in normal listeners. This suggests that masked thresholds in impaired listeners can be viewed as a combination of the masking expected in normal listeners and the masking represented by the hearing loss. To test this hypothesis, we fitted simultaneous-masking data for five impaired listeners using recent models of combined masking. The data were levels of narrow-band noise maskers at several frequencies required to just mask 2000-Hz probes presented at 5 to 25 dB SL. For masker frequencies in regions of maximal abnormal upward spread of masking, the model correctly predicts that: (1) growth of masking is more gradual in impaired listeners, (2) the maximum difference in thresholds for normal and impaired listeners occurs when the normal listener's masked threshold is equal to the impaired listener's quiet threshold, and (3) the size of this difference increases with the degree of hearing loss. The last result suggests that sensorineural hearing loss can best be simulated with forward maskers. [Work supported by NIH.]

4:00

HH13. Modeling sensorineural hearing loss. I. Approach and application to frequency resolution. Larry E. Humes, Blas Espinoza-Varas, and Charles S. Watson (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

In the present paper we evaluate the extent to which auditory-processing deficits observed in listeners with sensorineural hearing loss can be predicted entirely on the basis of loss of hearing sensitivity. In this model, processing deficits in the impaired ear are considered to be equivalent to those observed in normal listeners when masked by spectrally shaped noise. For normal listeners with such a simulated hearing loss, performance on tasks with actual masking noises thus involves the detection of signals in the presence of two maskers. One masker is used to simulate the hearing loss while the other is incorporated in the particular paradigm under study. The masked thresholds from each masker can be combined using a masking-additivity rule similar to that of Lutfi [J. Acoust. Soc. Am. **73**, 262–267 (1983)]. This approach is used to predict published data from impaired ears on critical ratios, critical bandwidths, auditory filter shapes, masking patterns, and psychophysical tuning curves. For most of the paradigms studied, the psychoacoustic performance of the hearing impaired was predicted reasonably well with this approach. These results seem to suggest that the frequency-resolution deficits observed in the hearing impaired are primarily a consequence of altered hearing sensitivity. [Work supported by NINCDS.]

4:15

HH14. Modeling sensorineural hearing loss. II. Application to measures of temporal resolution. Blas Espinoza-Varas, Larry E. Humes, and Charles S. Watson (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Thresholds of hearing-impaired listeners in temporal-masking tasks were modeled by the performance of normal listeners with a noise-simulated hearing loss. This model predicts thresholds for such tasks by assuming the presence of two maskers; a spectrally shaped continuous masker used to simulate the hearing loss and the masker used in the temporal-masking task. The combined effect of the two maskers was predicted using a masking-additivity rule similar to that of Lutfi [J. Acoust. Soc. Am. **72**, 262–267 (1983)]. Masked detection thresholds from impaired listeners and normal listeners with simulated hearing loss were examined for various temporal-masking paradigms. Data from the literature were examined for each of the following: (1) detection thresholds in continuous and modulated broadband noise; (2) growth of forward masking with masker level; and (3) decay of forward masking following masker offset. The masking-additivity approach predicted well the temporally masked detection thresholds for noise-masked normal listeners with simulated hearing loss, but underestimated the masked thresholds of the hearing impaired by as much as 15 dB. [Work supported by NINCDS.]

THURSDAY AFTERNOON, 14 MAY 1987 REGENCY BALLROOM A & B, 1:00 TO 4:29 P.M.

Session II. Speech Communication VIII: Speech Processing, Speech Synthesis, and Intelligibility

Mohan M. Sondhi and Shirley A. Steele, Co-Chairmen
AT&T Bell Laboratories, Murray Hill, New Jersey 07974

Chairmen's Introduction—1:00

Contributed Papers

1:05

II1. The design of a speech analysis workstation. John M. Crump (Kay Elemetrics Corporation, 12 Maple Avenue, Pine Brook, NJ 07058)

The recent availability of general purpose digital signal processing chips, inexpensive digital memories, and personal computers has opened the door to the development of a powerful workstation designed for the analysis of speech. A system can be developed with the advantages of a spectrograph (e.g., Sona-Graph), an oscillograph (e.g., Visicorder), a feature extractor (e.g., Visi-Pitch), and a general purpose computer (e.g., VAX with DSP software). This paper describes the development and capabilities of a standalone system based on a common microprocessor, powerful digital signal processing circuits, high-resolution graphic displays and high-speed DMA capabilities. The digital signal processing chip selected was the 32020 from Texas Instruments. Special high-speed DMA chips were used to facilitate high-speed data transfer between the different system modules (A/D to memory, memory to DSP circuits, DSP circuits to graphic circuits, and DSP circuits to printer). These DMA chips allow a 2-Mbyte/s transfer rate. The system management is performed by a Motorola 68000 and the system architecture has been defined to include up to 8 Mbytes of RAM. This system performs all the desired functions of a speech workstation and could provide a common platform for speech analysis research.

1:17

II2. Vocal jitter and shimmer measurements with a personal computer. David G. Drumright (Department of Mechanical Engineering, 22 Hammond Building, Pennsylvania State University, University Park, PA 16802), J. Anthony Seikel, and Kim A. Wilcox (Department of Speech–Language–Hearing, University of Kansas, 2101 Haworth, Lawrence, KS 66045)

Vocal fundamental frequency is one of the most widely analyzed components in speech acoustics. As such, there is a need for an efficient and accurate means of producing fundamental frequency data. This paper describes a series of machine language and basic routines that yield average fundamental frequency, vocal jitter, and vocal shimmer measures with a resolution sufficient for most research purposes. The algorithm permits the user to determine major period divisions through an efficient interactive graphic process. The microcomputer is then responsible for computing accurate period and intensity data for each cycle. In the present installation, the accuracy of these procedures is limited only by the 20 000-Hz maximum sampling rate of the hardware. In addition to the $f0$ analysis routines, programs for high-speed digital sampling, waveform editing, and waveform manipulation have been developed for use with PC-compatible computers. Documentation of the installation, use, and accuracy of the routines will be provided. [Work supported by NINCDS and U. of Kansas General Research Fund.]

1:29

II3. Some results of speech analysis using a parallel image processor. John F. Hemdal (Department of Electrical Engineering, The University of Toledo, Toledo, OH 43606)

Special purpose parallel pipelined image processors are not only faster for images than general purpose computers but also allow one to investigate the applications of the latest image processing algorithms and techniques to continuous speech. This is particularly valuable when trying to automate the spectrogram reading capabilities of Victor Zue [Proc. IEEE 73, 1602–1615 (1985)]. Several sentences of continuous speech were sampled and converted to images using the FFT. These images were analyzed on a parallel cellular image processor called the Cytocomputer™. Suitable structuring elements were selected to smooth the inherently bumpy FFT spectra and to remove noise. Several of the distinctive features of speech are described in mathematical morphology terms and these features were tracked. Specifically, stop consonants, fricatives, vowels, and nasals were located and labeled. Examples of intermediate image transformations and resultant labeling were recorded by photographic process. [Work supported by Ohio Board of Regents.]

1:41

II4. ARMA parameter estimation of speech. John J. Wygonski and Guy Sohie (Department of Electrical and Computer Engineering, Arizona State University, Tempe, AZ 85287)

Autoregressive moving average (ARMA) or pole-zero modeling techniques are known to provide better spectral estimation capabilities than autoregressive (AR) or all-pole methods such as linear prediction for nasalized speech and speech corrupted by noise. This paper describes the results of applying two ARMA modeling techniques to synthesized and natural segments of noisy and clean speech. The overdetermined normal equation (ODNE) and the extended-order singular value decomposition (SVD) methods of ARMA parameter estimation are compared to the covariance method of linear prediction and are evaluated using parametric distance measures in the cepstral domain. Use of the AR part of the ARMA model to estimate the envelope of noisy speech is demonstrated. Coefficient transformations among ARMA, AR, and cepstral parameters are also discussed.

1:53

II5. A codebook of articulatory shapes. M. M. Sondhi, J. Schroeter, and J. N. Larar (AT&T Bell Laboratories Murray Hill, NJ 07974)

The acoustical properties of a vocal tract depend on its shape which, at frequencies below about 4 kHz, is adequately described by the area function (i.e., cross-sectional area as a function of position from glottis to lips). Obtaining even this simplified description of the tract is a difficult problem. The solution that is proposed is to construct a linked codebook of area functions and corresponding LPC vectors. Then, a given LPC vector is transformed to an area function by finding the closest LPC vector in the codebook and accessing the linked area function. The authors have constructed just such a codebook (to our knowledge, the first). At present, it has about 1500 shapes clustered from a training set of about 10 000 shapes generated by a vocal tract model [P. Mermelstein, J. Acoust. Soc. Am. 53, 1070–1082 (1973)]. Distance between area functions is defined as the Itakura distance between the corresponding LPC vectors. Handling the unique problems arising from this definition of distance and from the large size of the training set will be described. The plan for refining and enlarging the codebook will also be discussed.

2:05

II6. Statistical representation of word-initial obstruents: General considerations. Gary Weismer, Karen Forrest, and Paul Milenkovic (Speech Motor Control Laboratories, University of Wisconsin—Madison, Madison, WI 53705-2280)

The description of obstruent spectra has typically been done in categorical terms [e.g., S. E. Blumstein and K. N. Stevens, J. Acoust. Soc. Am. 66, 1001–1017 (1979)]. Whereas this may be satisfactory for some purposes, the categorical approach is not helpful in cases where obstruent sounds are produced that are not easily described according to the normal set of segmental contrasts. In particular, many dysarthric and apraxic subjects produce stops and fricatives that give the perceptual impression of being "between" the normal segmental categories. The purpose of this paper is to describe some of the general considerations that led us to develop measures capable of indexing spectra associated with such indeterminate obstruent productions. A statistical representation of obstruent spectra will be described that is similar to certain approaches to describing a more general class of impulse noises [J. Erdreich, J. Acoust. Soc. Am. 79, 990–998 (1986)]. [Work supported by NIH Awards NS13274 and NS20976.]

2:17

II7. Statistical representation of word-initial obstruents: Adult data. Karen Forrest, Gary Weismer, and Paul Milenkovic (Speech Motor Control Laboratories, University of Wisconsin—Madison, Madison, WI 53705-2280)

The utility of a statistical representation of word-initial obstruents (Weismer *et al.*, previous paper) was assessed for the speech of normal, young adults. Ten subjects repeated monosyllabic words, where the initial consonant was a voiced or voiceless stop or fricative, in a carrier sentence. Each place of obstruent articulation was paired with at least two vowels and each target word was repeated six times. The target words were digitized and the first four moments (mean, variance, skewness, and kurtosis) were calculated for successive 10-ms spectral slices, beginning with the onset of the obstruent and continuing through the third cycle of the vowel. Moments were calculated from linear and bark transformed spectra. Three of the moments (mean, skewness, and kurtosis) provide a good description of the shape and center of the spectral slices. These moments were plotted in three-dimensional space to determine whether a unique, three-dimensional pattern could be defined for each place of articulation. Discussion will center on the use of these moments, as they define a three-dimensional space, to quantify acoustic events associated with obstruent production. [Work supported by NIH Awards NS13274 and NS20976.]

2:29

II8. Graphics editor for speech synthesis experiments. Andrew Helck, Ian Coville, and Noriko Umeda (Institute for Speech and Language Sciences, 10 Washington Place, New York University, New York, NY 10003)

A graphics editor of acoustic parameters for a speech synthesizer [an assembly language version of the software synthesizer by D. Klatt, J. Acoust. Soc. Am. Suppl. 1 **68**, S18 (1980)] has been written. The program provides an easy selection of various editing functions on the screen. It gives users visual patterns of time-varying acoustic parameters and the time relationship among parameters (such as that among fricative, hiss, and voice amplitude for stop consonants). Variable parameters are grouped into pages according to function, and simple key presses on the terminal allow users to select pages, parameters, and editing functions for the parameter. Editing functions are: moving the cursor, inserting points for linear interpolation, changing parameter values on curves, setting a mark for playing the designated portion, etc. In addition, there are commands to assign a set of parameter values at a specified point of time from a table of phoneme definitions. The editor has been used to synthesize short sentences and phrases of about 1s and has proven extremely useful. [Work supported by NSA.]

2:41

II9. The shape and alignment of rising intonation. Shirley A. Steele and Mark. Y. Liberman (AT&T Bell Laboratories, 2D 519, 600 Mountain Avenue, Murray Hill, NJ 07974)

Factors that affect the location of the nuclear stress $F0$ peak in statements have been described elsewhere, and the importance of correct peak placement in text-to-speech synthesis has been demonstrated [Steele (1986); Liberman and Steele (1986)]. This paper describes the temporal characteristics of the rising intonation pattern at the end of yes/no questions. In particular, certain questions of interest for text-to-speech synthesis have been asked: What parameters are needed to describe the shape of this $F0$ contour and its alignment with segments of the text? Specifically, how do the shape and alignment of the contour change with variations in the type and timing of post-nuclear segments and with changes in speech rate and pitch range?

2:53

II10. Implementing prosodic phrasing for an experimental text-to-speech system. Joan Bachenko, Eileen Fitzpatrick, and John Lacy (AT&T Bell Laboratories, Murray Hill, NJ 07974)

While text-to-speech systems tend to perform well on word pronunciation, they fall short when it comes to providing good prosody for complete sentences. An experimental text-to-speech system that uses a natural language parser and prosody rules to determine prosodic phrasing for English input to text-to-speech synthesis will be described. Building on information from the syntax tree, the prosody rules specify the location and relative strength of prosodic phrase boundaries; these specifications are then used to dictate modulations in pitch and timing for the Olive–Liberman synthesizer [J. P. Olive and M. Y. Liberman, J. Acoust. Soc. Am. Suppl. 1 **78**, S6 (1985)]. Two important assumptions motivate the prosody rules. First, constituency below the level of the sentence is the crucial determinant for boundary location; i.e., boundary location is influenced by noun, prepositional, and adjective phrases, but not by clauses or verb phrases. Second, the relative strength of boundaries is determined by balancing prosodic phrases around a verb (or predicate) according to length, adjacency to a syntactic head, and the subject–predicate relation. The ability to predict phrasing reliably allows the system to use an expanded pitch range and insert pauses appropriately.

3:05

II11. Nasal vowel study: Formant structure, perceptual evaluation, and neural representation in a model of the peripheral auditory system. Yan Ming Cheng and Bernard Guérin (Institut de la Communication Parlée, UA 368 au CNRS, 46 avenue Félix Viallet, 38031 Grenoble Cedex, France)

In this study, the focus is on the search for a model that is articulatorily possible and perceptually plausible. A high-quality nasal sound synthesis structure in the acoustic domain, with a parallel formant synthesizer, that includes nasal articulation effects and defines independent acoustic parameters which are the basis of perceptual studies, has been obtained. A pole–zero–pole structure combined with a bank of formants is proposed as a high-quality nasal vowel synthesizer. A set of perceptual tests is conducted to test nasality variation with nasal synthesis eigenparameters. The results of tests are used to establish a new model describing [+ nasal]. Based on this model, the nature of the French nasal vowel and its distributional properties are discussed. Finally, nasality variation is tested at the auditory-nerve level with the help of a model of the peripheral auditory system. The neural firing pattern and the synchronization index pattern are discussed in terms of the nasality model established.

3:17

II12. Calibration and the comprehension of synthetic and natural speech. Bonnie L. Zeigler, George J. Boggs, and Leah S. Kaufman (Behavioral Sciences Department, GTE Laboratories, 40 Sylvan Road, Waltham, MA 02154)

Investigations of "calibration" (the relationship between knowledge and the self-assessment of knowledge) have found that humans reading expository text cannot accurately judge their ability to draw inferences from the material read. Whether low calibration was related to the inference process or to the act of reading by presenting passages aurally, as either natural or phoneme-synthesized speech is examined. Sixty-four adults, divided into natural and synthetic-speech groups, heard 15 passages, with half the subjects permitted to listen twice. After each passage, subjects rated their comprehension, then judged the veracity of an inference drawn from the text. Calibration quotients, derived from the ratings and performance scores, were significant for both synthetic and natural groups. Performance, ratings, and calibration were slightly higher for natural versus synthetic subjects, but the difference was reliable only for confidence ratings. Thus previous low calibration may be more related to presentation mode than to inference processes. Better calibration for listeners may be due to a closer relationship between inferences and listening, which encourages extraction of themes as opposed to detail, than between inference and reading.

3:29

II13. Children's perception of synthetic speech produced by rule. Beth G. Greene (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Several experiments examined the perception of synthetic speech by school-age children ranging in age from 5- to 11-years old. These tasks all used synthetic speech produced by commercially available text-to-speech systems. In tasks involving free and ordered recall of lists of words, 8-year-old children recalled naturally produced items better than synthetically produced items even when the items were equated for familiarity and intelligibility. Other tasks included children's responses to spoken commands and sentence recall using meaningful and semantically anomalous sentences. As in our previous studies, performance was better for natural speech than synthetic speech, paralleling the general findings obtained with adults in our earlier studies. Synthetic speech, even when the speech is highly intelligible, may interact with task requirements to produce significant decrements in performance when children engage in complex cognitive tasks. [Supported by NINCDS.]

3:41

II14. Some relationships between preference and intelligibility among synthetic voices. John S. Logan (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

The relationship between preference and intelligibility among synthetic voices was examined. Preferences among the speech produced by three text-to-speech systems were assessed using an A/B paired comparison procedure, and the intelligibility of the synthetic voices was measured using a transcription task. The text-to-speech systems tested were DEC talk, Prose 2000, and Infovox. Phoneme specific sentences served as stimuli and enabled a preliminary examination of the relationship between intelligibility and preference as a function of specific phoneme categories. Results indicated a close relationship between mean intelligibility and preference for a voice. In some cases, preference was found to vary with the intelligibility of individual classes of phonemes, accounting for some of the variability associated with preference decisions. These results will be discussed in terms of previous research involving preference and intelligibility. [Work supported by NIH and USAF.]

3:53

II15. Some measures of intelligibility of natural and coded speech for native and non-native speakers of English. Kazunori Ozawa (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405 and C&C Information Technology Research Laboratories, NEC Corporation, Kawasaki 213, Japan) and John S. Logan (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Segmental intelligibility of both natural and coded speech was measured using the modified rhyme test (MRT). Pitch predictive multipulse excited speech coding (MPC) [K. Ozawa and T. Araseki, Proc. ICASSP86 33.3, 1689–1692 (1986)] was used to produce high-quality coded speech at 8 kb/s. As a reference, μ-law PCM was also used. To investigate not only perceptual differences between natural and coded speech but also the differences due to linguistic knowledge and experience with English, both native and non-native speakers of English were used as subjects. All of the stimuli were bandlimited through a low-pass filter with 4.5-kHz cutoff frequency and sampled at 10-kHz sampling frequency. Closed format MRT scores were obtained to measure the intelligibility for three conditions, natural speech, μ-law PCM and 8 kb/s MPC. Detailed analysis of the results of the intelligibility test will be presented. Differences between the intelligibility scores of native and non-native subjects and the relationship between these scores and their linguistic experience will also be discussed.

4:05

II16. Speech enhancement and the instantaneous phase signal. S. J. Walsh and P. M. Clarkson (Institute of Sound and Vibration Research, University of Southampton, Southampton, England)

An approach to speech enhancement based on a representation of the speech signal by an envelope and instantaneous phase model is proposed. Envelope modification for the removal of room reverberation and noise from signals has already been studied and found to produce signal-to-noise ratio but not intelligibility improvements. The envelope alone does not account sufficiently for the detailed structure of speech. A noisy instantaneous phase has been found to have a perceptually degrading effect even with a perfectly restored envelope. Thus it is proposed to enhance speech by also replacing the measured instantaneous phase signal (in a number of contiguous frequency bands) by a modeled version. To determine the characteristics of the instantaneous phase function both synthetic and recorded speech have been studied. A simple model for the instantaneous phase of speech is proposed and an analytical solution is developed relating the digital model of speech production to an envelope and instantaneous phase model. Enhancement of a noisy instantaneous phase function can be achieved by using a compensating frequency shift, low-pass filtering and linear prediction.

4:17

II17. Adaptive speech intelligibility testing for communication systems. D. L. Hogan, C. C. Gumas, C. R. Moore (SPARTA, Inc., 7926 Jones Branch Drive, McLean, VA 22102), B. T. Oshika (SPARTA, Inc., 2560 Ninth Street, Berkeley, CA 94710), S. L. Hamlet (Department of Hearing and Speech Sciences, University of Maryland, College Park, MD 20742), and D. B. Pisoni (Speech Research Laboratory, Indiana University, Bloomington, IN 47405)

An automated adaptive interactive speech intelligibility test was implemented on an IBM-PC. Subjects hear digitized short sentences such as "grass is green" or "escalators have doors" over headphones and must key in a T or F response for each sentence. Quality of speech stimuli is controlled by different iterations of a degrading transfer function before playback. The level of degradation presented to the listener in successive trials is dependent on the just-previous pattern of accuracy in subject response. The test run is programmed to converge to that level of stimulus degradation at which the subject responds at a selected % correct criterion. The test is repeated for multiple % correct criteria, to permit not only assessment of a communication system under good intelligibility conditions but also measurement of sensitivity to reduced intelligibility. The approach permits flexible manipulation of parameters such as speech stimulus material, type of stimulus degradation, response time measurement, etc. The test is under computer control from initial instructions to the subjects to final summary of responses. The automated adaptive procedure has been evaluated for 100 normal hearing subjects. [Work supported by ESD, AF Systems Command.]

THURSDAY AFTERNOON, 14 MAY 1987 — CELEBRATION HALL, 1:15 TO 4:45 P.M.

Session JJ. Bioresponse to Vibration III and Physical Acoustics VI: Biophysical Acoustics: Absorption in Biological Media II

William D. O'Brien, Jr., Chairman

Department of Electrical and Computer Engineering, University of Illinois, 1406 W. Green Street, Urbana, Illinois 61801

Contributed Papers

1:15

JJ1. Absorption of ultrasound in denatured proteins. Frederick W. Kremkau (Ultrasound Center, Bowman Gray School of Medicine, Wake Forest University, Winston-Salem, NC 27103)

Absorption data have been obtained on nine globular proteins in aqueous solution denatured with 6M guanidine hydrochloride at neutral pH. The hypothesis that there is an underlying absorption behavior with respect to molecular weight similar to that for dextran [Kremkau and Cowgill, J. Acoust. Soc. Am. 76, 1330–1335 (1984)] was tested. If the scatter in previous data for native proteins is due to structure, denaturing would draw the points toward the underlying value. However, the scatter exists in the denatured protein absorption values as well. Various solvent contributions to absorption of the denatured proteins may cause the scatter. A second hypothesis was tested: secondary structure contributes to absorption in globular proteins but tertiary structure reduces this contribution because of solvent shielding [Kremkau and Cowgill, J. Acoust. Soc. Am. 77, 1217–1221 (1985)]. This hypothesis predicts that solvent buffer ions would increase absorption to a greater extent in denatured form (with many more side chains exposed to the solvent) than in native form. Such behavior was seen in some, but not all of the proteins. [Work supported by NCI, DHHS.]

1:30

JJ2. Ultrasonic propagation properties of excessively high fatty rat livers determined with the scanning laser acoustic microscope. William D. O'Brien, Jr. (Department of Electrical and Computer Engineering, University of Illinois, 1406 W. Green Street, Urbana, IL 61801) and John W. Erdman, Jr. (Food Science Department, University of Illinois, 506 S. Goodwin, Urbana, IL 61801)

The effects of 1% dietary orotic acid on the ultrasonic propagation properties of rat livers were examined. In rats, dietary orotic acid exerts several effects on lipid metabolism; its overall consequence is that excessively high fat concentrations are built up over a short period of time, thus making this an ideal model to study the ultrasonic propagation properties as a function of fat in liver. Over a 16-day period of time in which the rats were on the orotic acid diet, lipid concentrations increased from a normal range of 2%–4% to the lower 20's%. Additionally, water concentration decreased from normal values of approximately 70% to approximately 50%. Total protein concentration decreased slightly from a normal of 17%–20% to 11%–16%. The rat liver weight increased from approximately 11 g to around 20 g. Ultrasonic attenuation coefficient and speed were assessed with the scanning laser acoustic microscope at 100 MHz. Strong correlates of ultrasonic speed with both water concentration and fat concentration in the liver were observed. Multivariant linear regression was used in the analysis of covariance to fit the least-squares estimates to the linear regression model.

1:45

JJ3. Ultrasound absorption and attenuation loss in liver in the finite amplitude range. F. J. Fry (Labsonics, Inc., Mooresville, IN 46158), K. Dines (XData, Indianapolis, IN 46206), and M. Etchison (I. C. F. A. R., Indianapolis, IN 46204)

The ultrasound absorption coefficient in fresh excised rat liver has been obtained at 1-MHz fundamental frequency in the intensity range (SPTP) from low intensity (50 W cm^{-2}) to high intensity (800 W cm^{-2}). At atmospheric pressure the pressure absorption coefficient starts at 0.035 (50 W cm^{-2}) and rises to 0.090 (800 W cm^{-2}). Theoretical calculations over this same intensity range show agreement in the shape of the absorption coefficient curve as a function of intensity up to approximately 450 W cm^{-2}. Above this intensity the experimentally derived values flatten off and slightly decline with increasing intensity. The theory shows no such characteristic. The absorption coefficient was also measured in the same intensity range at a pressure of 350 lb $in.^{-2}$. (This pressure provides a bias for 193 W cm^{-2} at 1 MHz.) There is a small (10%) and statistically significant reduction in the absorption coefficient in the intensity range from 200–500 W cm^{-2} but below and above this range there is no statistically significant difference in the absorption values from those obtained at atmospheric pressure. Attenuation losses show the same characteristic charges with intensity as does the absorption coefficient but the total attenuation loss is approximately twice that due to absorption alone. [Work supported by NIH grant 5R01 CA 41073-05.]

2:00

JJ4. Ultrasonic absorption in mammalian ovaries. K. I. Carnes and F. Dunn (Bioacoustics Research Laboratory, University of Illinois, 1406 W. Green Street, Urbana, IL 61801)

The human ovary, as well as that of other mammalian species, has the opportunity to receive ultrasonic exposure during diagnostic and therapeutic procedures. However, the ultrasonic absorption behavior seems never to have been investigated. Such information is considered essential for both safety and treatment planning. The ovary is a dynamic organ exhibiting interspecies variation as well as variation among the intraovarian structures, viz., the follicles, corpora lutea, etc. The *in vitro* ultrasonic absorption coefficient was determined, by the transient thermoelectric method, for ovaries of bovine, canine, feline, ovine, and murine species at 1 MHz. Similarities in values exist for intraovarian structures among the species measured. For example, of the stromal tissues, the outer cortex layer exhibits absorption coefficients approximately 50% greater than that of the inner medulla. The follicle absorption is the lowest and the corpora lutea values vary with the stage of development. These results are consistent with the macromolecular composition of these structures. [Work supported by the NIH.]

2:15

JJ5. Frequency spectrum shifts of ultrasound echoes as a function of depth in a female breast. Elizabeth Kelly-Fry (Indiana University School of Medicine, Wishard Memorial Hospital, Department of Radiology, Indianapolis, IN 46202), Steven T. Morris, Narendra T. Sanghvi (Labsonics, Inc., Mooresville, IN 46158), and Ernest L. Madsen (Department of Medical Physics, University of Wisconsin, Madison, WI 53706)

Knowledge of the changes in waveform characteristics as an ultrasound wave penetrates multiple layers of soft tissues within a breast can be important for both detection and diagnosis of breast pathologies by ultrasound mammography techniques. Using a spectrum analyzer system in combination with an automatic, bi-plane scanning, *B*-mode breast imaging unit, techniques were developed for recording the center frequency and bandwidth characteristics of signals received from the subcutaneous fat, mid-breast, retromammary, and pectoralis muscle regions of the breast of a post-menopausal subject. The same techniques were applied to a breast phantom that has internal architectural features and acoustic parameters (density, speed, attenuation coefficient) that are comparable to those of a normal breast. A single focus, 7.5-MHz transducer was used for most of the recorded data. Lower frequency transducers were also used. Spectral analysis data accompanied by relevant breast and phantom images will be presented. [Work supported by Indianapolis Center for Advanced Research and Labsonics, Inc.]

2:30

JJ6. Intense focused ultrasound for producing complex volume lesions in brain. Francis Fry, N. T. Sanghvi, Rich Morris (Labsonics, Inc., Mooresville, IN 46158), S. Griffith, and J. Hastings (I. C. F. A. R., Indianapolis, IN 46204)

The ultrasound dosage for producing single site focal lesions in the experimental animal brain has been well documented. Complex sheet lesions in white matter have been generated from a multiplicity of individual lesions at 1-MHz frequency without undesirable side effects in animal brains. At 4-MHz frequency, lesions in both white and gray matter involving a multiplicity of individual lesions have been generated without side effects in the animal brain. At 1-MHz frequency, the production of complex lesion shapes from a multiplicity of individual small lesions in both white and gray matter required establishing the lesion site intensity that would not initiate off primary site undesirable side effects. Intensities below 300 W cm^{-2} at the primary lesion site do not lead to such effects. A good working hypothesis for production of side effects involves microbubble formation at the primary lesion site. These microbubbles can be carried through the vascular circulation providing sites for cavitation formation outside the primary lesion site. It is now possible to prescribe primary lesion site intensities in brain in the frequency range from 1–4 MHz that negates the possibility of microbubble formation. [Work supported by Labsonics, Inc., Mooresville, IN.]

2:45

JJ7. Human gallstone dissolution accelerated by ultrasound application. Francis Fry, N. T. Sanghvi (Labsonics, Inc., Mooresville, IN 46158), Bryan Burney (St. Francis Medical Center, Beech Grove, IN 46107),

S. Griffith, J. Hastings, and T. D. Franklin (I. C. F. A. R., Indianapolis, IN 46204)

Gallstones in appropriate chemical species subject to ultrasound exposure show accelerated dissolution over that accruing to the chemical alone. *In vitro* studies using a human gallstone submerged with monooctanoin (a cholesterol stone solvent) in a latex envelope show dissolution of a stone mass of 0.50 g with 6 min of total ultrasound exposure at a SPTP intensity of 15 W cm^{-2}. Control stones in the monooctanoin show a weight loss of 0.05 g in the same overall treatment time. When a pressure of 350 lb in^{-2} is applied the stone dissolution rate declines by a factor of 3 from that at atmospheric pressure. The primary erosion mechanism is apparently of cavitation origin but a smaller but significant erosion acceleration occurs in the apparent absence of cavitation. A unique feature of this type of gallstone dissolution is the conversion of solid stone into a liquid state providing for ease of passage of gallbladder content in the clinical situation. That this accelerated dissolution occurs *in vitro* has been demonstrated on a series of pigs with human gallstones implanted in their gallbladder. [Work supported by Indiana Corporation for Science and Technology, I. C. F. A. R., and Labeco/Labsonics, Mooresville, IN.]

3:00

JJ8. A novel approach of generating various frequency bands using a wideband PVDF ultrasound transducer and its application for tissue characterization. N. T. Sanghvi, S. T. Morris, P. W. Wendt (Labsonics, Inc., 236 E. Washington Street, Mooresville, IN 46158), and E. Kelly-Fry (Indiana University School of Medicine, Wishard Memorial Hospital, 1001 West 10th Street, Indianapolis, IN 46202)

The technological advances in electronics in recent years have made it possible to manufacture polyvinylidene fluoride (PVDF) ultrasound transducers with sensitivity that could be routinely used in clinical instruments for ultrasound imaging and diagnosis of tissue. Last year, we have taken a noval approach and developed electronics to excite PVDF transducers with a sharp spike of voltage with controlled energy-frequency spectrum to generate and transmit a selected frequency band with its center frequency totally controlled by the excitation pulse and selected filter. With this approach four different center frequencies with approximate band width of 45%–50% is generated. This paper will describe the analytical model and compare theoretical data with the experimental results. As a result of this development a prototype device is fabricated to study various breast tissue types and differentiate them based on their scattering characteristics as a function of different frequency bands. ["Bi-Plane Scanning with a High Frequency, Multi-Frequency Ultrasound Instrument for Improved Differentiation of Malignant and Benign Breast Tumors," E. Kelly-Fry *et al.* submitted for publication in Radiology. J.]

3:15

JJ9. Acoustically enhanced thermal diffusion. Charles Thompson (University of Lowell, Department of Electrical Engineering, One University Avenue, Lowell, MA 01854)

The role of acoustic vibration in the enhancement of the thermal diffusion process near a solid boundary is discussed. Our purpose is to discuss the degree with which flow unsteadiness can be exploited for the purposes of transport augmentation. There are three factors that govern augmentation: the behavior of the Stokes layer; unsteady flow instabilities; and the geometry of the surface boundary. The interelationship of these three factors on the augmentation process will be discussed and preliminary results of our analysis presented.

3:30–4:45

Bull Session

THURSDAY AFTERNOON, 14 MAY 1987 — CANYON HALL, 1:30 TO 4:20 P.M.

Session KK. Musical Acoustics IV: Acoustics of Pianos II

Uwe J. Hansen, Chairman

Department of Physics, Indiana State University, Terre Haute, Indiana 47809

Chairman's Introduction—1:30

Invited Papers

1:35

KK1. Piano string excitation: Theory and reality. Donald E. Hall (Physics Department, California State University, 6000 J Street, Sacramento, CA 95819)

An idealized model of piano string excitation has been shown soluble by a combination of analytic and computer techniques. It is also useful to step aside from the mass of details this model generates, and discuss general physical arguments about what overall envelopes should limit the string vibration spectra produced. This also provides a framework for more careful consideration of the limits of validity of the theory. It is argued that the most important weakness is the non-Hookean nature of the hammer compliance, followed by string stiffness effects; possible ways of including these effects will be considered. Further laboratory measurements of string motion will be presented, especially bearing upon the question of whether striking the string a fraction $1/N$ of its length from one end does or does not cause mode number N to be missing from the spectrum. Measurements of radiated sound that would help clarify the role of the soundboard will also be considered.

2:05

KK2. Vibration analysis of a hammer–shank system. Hideo Suzuki (Acoustics Laboratory, Ono Sokki Company, Ltd., Shinjuku-ku, Tokyo, 163 Japan)

The transient motion of a hammer–shank system is mathematically analyzed using a modified form of the Myklestad method. The hammer is modeled as a mass–spring system with a nonconstant spring coefficient. The shank is divided into 13 elements, where each element is assumed to have a uniform cross section. The effect of the shank stiffness, the hammer velocity, and the initial deformation of the shank in the force–time pattern applied by the hammer to an ideally rigid string are investigated. The results show that the shank deforms significantly during the hammer–string contact period. It is also shown that the force–time pattern changes significantly when an initial deformation of the shank is introduced. Some of the theoretical results are compared with experimental results. (This study was conducted while the author was at CBS Technology Center, Stamford, CT 06905, from 1981 to 1985.)

2:35

KK3. Piano hammer: A nonlinear model compared to experiment. Xavier Boutillon (Laboratoire d'acoustique, Université Paris VI, T66 5ème étage, 4, place Jussieu, 75005 Paris, France)

This paper presents a nonlinear model for the hammer, its experimental determination, and comparisons between simulated and recorded movements of strings and hammer during their interaction. It is first shown how closely the impact velocity of the hammer is the only input parameter that governs the collision of the hammer with the strings and therefore the quality of the sound. However, this quality is highly dependent on the magnitude of the impact velocity. So, a nonlinear model is required to describe the hammer's behavior. It consists of the association of a mass with a nonlinear, hysteretic spring whose characteristics are experimentally determined under realistic playing conditions. Both nonlinearity and hysteresis are fit by analytical relationships. This model, when associated with a simple flexible string model, allows a numerical simulation of the movements of both hammer and strings. These are compared to the measured vibration of a string and to the measured acceleration of the hammer. In addition, results are presented regarding the efficiency of the hammer impact.

Contributed Papers

3:05

KK4. New method of measuring input admittance of stringed instruments. Gabriel Weinreich (Randall Laboratory, University of Michigan, Ann Arbor, MI 48109) and Levon Yoder (Department of Physics, Adrian College, Adrian, MI 49221)

When a stringed instrument, such as a violin, is subjected to an applied time-harmonic force and the resulting velocity of the bridge is measured, the quantity usually referred to as the "input admittance" in fact functions as an output admittance. Its value can be inferred by loading the bridge with a known external impedance and comparing the velocities with and without this loading. In our experiment, the external loading impedance is a known mass, and the applied force results from an irradiating acoustic field. By taking data with two different field configurations, we can independently determine the four elements of the two-dimensional input admittance matrix referred to the two possible polarization directions of the string vibration. We utilize the digital data acquisition and processing system that has been previously developed for radiativity measurements, in which an appropriate broadband stimulating signal allows a large spectrum of data to be obtained in a few seconds. Some results will be shown. [Work supported by NSF.]

3:20

KK5. Boundary conditions and violin vibrations. Kenneth D. Marshall (The Uniroyal Goodrich Tire Company, Research and Development Center, Brecksville, OH 44141)

In this paper the vibrational behavior of a violin is examined under a variety of boundary conditions; held by the player, freely suspended, pinned supports, simple supports, clamps applied to the neck and corpus, and supported by sandbags. It was found that three different players imposed very similar constraints on the instrument, with the principal effect being a significant increase in the modal damping as compared to the free vibrational response of the violin. The effect of the player on the modal frequencies was negligible. It was also found that resting the instrument on a piece of foam rubber or the use of sandbags under the neck and the tail of the instrument provided a good first approximation to the influence of the player. The other boundary conditions that were investigated caused significant deviations from the normal response of the instrument, thus indicating that these mounting techniques should not be used to measure the vibrational behavior of a violin or other stringed instrument.

3:35

KK6. Iranian scale of music can be considered as a universal scale. M. Barkeshli (Department of Physics, Tehran University, Tehran 13887, Iran)

Iranians divide the major tone in two: Limma (L) and a comma (C). Safi-yud-din (the Iranian savant virtuoso from the 13th century) was the pioneer of a school that propagated the systematic theory, taking L as unity for intervals, and divided the tone 9/8 into L L C. We see this division in the tablature of tunbur Khorasan, a typical instrument used in Iran and neighboring countries (description by Farabi, the Iranian philosopher and musicologist from the 10th century). Our researches on measuring the intervals of contemporary Iranian scale ["La Gamme de la Musique Iranienne," Télécommunication, Paris **5**, 5 (1950)] showed that there are also other divisions L C L and C L L in Iranian music. With this arrangement, the octave is divided into 27 intervals or 28 degrees. This scale can be considered as a universal scale because it contains all elements of harmonic and melodic scales. In fact the interval L + C, the characteristic of oriental music, is equal to 16/15 with the difference 1.0011292, being less than 1/50 of a semitone, a difference nondiscernable by the normal ear [V. O. Knudsen, *Architectural Acoustics*]. Also with the same difference $10/9 = 2L$, $5/4 = 4L + C$, and so on. [Work supported by ASI and PSI.]

THURSDAY PM

3:50

KK7. An exploration of the Chinese pure-tone system. Hai-si Pan and Yu-An Rao (Institute of Physiology, Academy of Sciences, Shanghai, People's Republic of China and Department of Psychology, University of California, Los Angeles, Los Angeles, CA 90024)

After 300 years of effort in many countries aimed at resolving the inharmonicity of the 12-interval equal-tempered scale, a pure scale of 53 tones was proposed in 1816–1890, but it can not be applied usefully owing to its very great complexity. A survey of ancient Chinese musical literature and instruments and deductions from "Jing-Fang's 60 Notes" of the Han Dynasty (漢 , 209 BC–220 AD) and Emperor Liang-Wu's "Four Pass" scale (502–519 AD), led us to propose a 23-tone scale. Its error curve corresponds precisely with that of the 53-tone pure scale. The principle of Schisma is used for solving problems such as (a) that of the inharmonicity between Gong (do) and Jiao (mi), i.e., the major third; and (b) that the major and minor chords coexisted in an octave. We have derived a pure-temperament scale of 23 tones that can be modulated for every tonic and that is consistent with the modern pure scale of 53 tones. We show that this 23 unequal-interval pure scale is consistent with the intervals which were obtained by measurement from a stone flute made in the Qing Dynasty (清 , 1644–1908 AD).

4:05

KK8. A comparison of the musical scales of an ancient Chinese Bronze bell ensemble and the modern bamboo flute. Yu-An Rao (Institute of Physiology, Academy of Sciences, Shanghai, People's Republic of China), Edward C. Carterette (Department of Psychology, University of California, Los Angeles, CA 90024), and Yu-Kui Wu (Institute of Acoustics, Nanjing University, People's Republic of China)

We compare over a two-octave range the tonal, interval, and scale relationships of an ancient Chinese bell ensemble of the Zhou Dynasty and a modern bamboo flute. By using subjective pitches, a tonal system with simple harmonic partials (the flute) could be compared with a tonal system having complex inharmonic partials (the bell). We argue that the underlying musical scales of the bell ensemble and the flute are closely related and lie between an unequal-interval pure system scale and the just-intonation scale. There is some evidence that at least an interval of 60 or 66 cents figures in the flute scale, that both flute and bell scales include a 90-cent interval and a true (1200) cent octave, and that neither scale includes a 100-cent interval. Since there is a true octave the scale cannot be a cycle of fifths, and since there is no equal-temperament half-tone (100 cents), the scales cannot be of equal temperament. We conclude that the flute and Zhou bell scales are very similar if not the same. Apparently strong cultural traditions and human perceptual constancies united to sustain a common flexible musical scale during twenty-four hundred years.

THURSDAY AFTERNOON, 14 MAY 1987 — MT. McKINLEY 1:30 TO 4:35 P.M.

Session LL. Underwater Acoustics VI: Matched Field Signal Processing—Part II: General Signal Processing

John M. Ozard, Chairman
Defence Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada

Chairman's Introduction—1:30

Contributed Papers

1:35

LL1. Bottom effects on matched-field localization performance in the North Pacific. R. G. Fizell and M. B. Porter (Code 5120, Naval Research Laboratory, Washington, DC 20375)

PACIFIC ECHO was an experiment conducted jointly by the Naval Research Laboratory (NRL) and Defence Research Establishment, Pacific (DREP) in May–June 1986. Vertical array measurements of a 15-Hz cw signal projected by the NRL MK-VI source, towed at a depth of 100 m, were taken on the 675-m-long DREP array at both thinly (<50 m) and thickly (>500 m) sedimented areas. The array was located in separate cases in the direct path and the shadow zone, and the top sensor was at a depth of approximately 350 m. Matched field processing between theoretical and measured fields was applied to this data set in order to perform range-depth localization. Both a linear (Bartlett) and a nonlinear, minimum-variance (Capon) estimator were employed using theoretical fields generated by a normal modes program. The bottom model plays an important role in the shadow zone where acoustic energy is received predominantly via bottom interacting paths. The thin sediment site, in turn, is especially interesting because of the interaction with the elastic subbottom. The effect of differences in measurement site (thin or thick sediment), source–received separation, and processor estimator on performance of the matched field processor are discussed.

1:50

LL2. Broadband acoustic source localization. A. Tolstoy and M. B. Porter (Code 5120, Naval Research Laboratory, Washington, DC 20375-5000)

This paper examines aspects of matched field processing for locating a broadband source in deep water. Concentration is on nearby sources (within 5 km of a vertical receiving array), on frequencies from 100 to 300 Hz, and on deterministic, simulation "data" only. The noise field has been modeled as Gaussian, uncorrelated, white noise while the acoustic field has been modeled by the fast field program, with subsequent processing by conventional and maximum likelihood (ML) methods to generate range-depth ambiguity surfaces (AMSs). The effects of mismatch in the description of sea surface roughness have been examined and it has been found that it can lead to significant errors in peak location on an AMS, particularly at the higher frequencies. However, incoherent frequency averaging of AMSs can, in the presence of relatively small mismatch, overcome the effect of such mismatch, even though almost all the component frequencies show errors in their predictions. Moreover, consideration of range averaging plus frequency averaging can, under such conditions, improve those cases not resolved by frequency averaging alone. It is hypothesized that frequency and/or range averaging will help only when the mismatch is sufficiently small that the AMSs have at least local maxima near the true source location.

2:05

LL3. Source localization in almost-stratified waveguides. E. C. Shang (Department of Computer Science, Yale University, New Haven, CT 06520)

In our previous papers [E. C. Shang, C. S. Clay, and Y. Y. Wang, J. Acoust. Soc. Am. **78**, 172 (1985); J. Acoust. Soc. Am. **77**, 1413 (1985)], source localization techniques based on mode filtering have been developed for stratified waveguides. In this paper, some problems of source localization in almost-stratified waveguides are discussed. Specifically, the Prony method is proposed for source bearing in range-dependent waveguides. It is found that the Prony method only requires a "local almost-stratified" condition, which means that within the data sampling aperture length the field can be treated as adiabatic modes. For most of the practical interesting cases, this condition can be satisfied quite easy.

2:20

LL4. Extraction of average under-ice reflection amplitudes and phases with matched field processing. E. Livingston and O. Diachok (Naval Research Laboratory, Code 5120, Washington, DC 20375-5000)

Average low-frequency under-ice reflection amplitudes and phases in the central Arctic were extracted from long-range (259 km) signals from fixed cw sources detected on a long (1 km) vertical array (the FRAM IV experiment) using conventional [H. Bucker, J. Acoust. Soc. Am. **59**, 368 (1976)] and maximum likelihood [R. Fizell, J. Acoust. Soc. Am. (to be published)] matched field processing methods. Theoretical computation of amplitudes and phases for all assumed ranges and depths were based on the Porter–Reiss [J. Acoust. Soc. Am. **77**, 1760–1767 (1985)] normal mode code. Under-ice reflection amplitudes and phases were incorporated into the propagation code and varied iteratively to achieve maximum signal gain and minimum range and depth errors. The resultant best data-fitting amplitudes and phases will be compared with expectations based on under-ice scattering theories and laboratory-scale model experiments.

2:35

LL5. Ventriloquism and spurious sound sources in underwater acoustics. Ian Roebuck (Admiralty Research Establishment, Portland, Dorset DT5 2JS, England)

With the increased use of active noise control has come wider awareness that the source distributions generating prescribed sound fields are not unique. In particular, the possibility of precisely reproducing the field due to a time-varying monopole by a multipole source (of infinite order) located elsewhere, has been established. In this paper, the fundamental physical limitations of carrying out such an "underwater ventriloquism act" in practice—developing a "constructive" algorithm for the various multipole coefficients and criteria for truncating the process are examined. It is shown that this is closely related to earlier ideas on the effective size of "point" sources [I. Roebuck, J. Acoust. Soc. Am. Suppl. 1 **69**, S87 (1981)] and further that acoustic efficiency limits the extent to which the transmitting multipole elements can be compacted without destructive mutual interaction. The manner in which the greater effective size of the "spurious" multipole source limits the potential for deception in the presence of varying boundaries is also analyzed.

2:50

LL6. Space-time processing, environmental–acoustic effects. W. M. Carey (Naval Underwater Systems Center, New London, CT 06320) and W. B. Moseley (Naval Ocean Research and Development Activity, NSTL, MS 39529)

The processing of acoustic waveforms by arrays requires an understanding of the temporal and spatial characteristics of signal and noise fields. Temporal and spatial processing schemes are analogous transforms that can employ a variety of windows (such as Hann, Hamming, etc.). However, the ocean environment is a filter that introduces variability to a signal in both spatial and temporal domains. This randomness is superimposed on an ambient sound channel characteristic. In the case of static source and receiver combinations, the limits on horizontal broadside array resolution are due to volume scattering and surface scattering as long as the time scale is less than the signal correlation time. However, in the case of a moving source–receiver, the temporal and spatial scales are coupled through the sound channel characteristic and the fluctuation effects due to multipath or modal variations must also be considered. This paper reviews fundamental environmental effects and their influence on arrays in the deep ocean sound channel. [Work performed while at NORDA.]

3:05

LL7. Passive synthetic arrays. W. M. Carey (Naval Underwater Systems Center, New London, CT 06320)

Passive-synthetic apertures [W. Carey and N. Yen, J. Acoust. Soc. Am. Suppl. 1 **75**, S62 (1984)] were formed with experimental towed hydrophone data in a sound channel that supported RR and RSR transmission. These apertures were formed with lengths up to 95λ with coherent temporal processing gains approaching 0.75 of theoretical. These results and those of previous investigators [R. Fitzgerald, J. Acoust. Soc. Am. **60**, 752–753 (1976); R. Williams, J. Acoust. Soc. Am. **60**, 60–73 (1976)] indicate that synthetic apertures can be formed by the coherent summation of the phase-corrected summation of either hydrophone or subaperture beams over successive time samples when the synthetic aperture length is less than the spatial coherence length and the processing time is less than the temporal coherence length. The evaluation of synthetic apertures requires comparisons with conventional and other high-resolution techniques. Comparisons between conventional array processing and high-resolution techniques [maximum entropy (ME) and maximum likelihood (ML) methods] are performed by use of the analytical expressions developed by A. T. Parsons (AUWE, TN 700/83) for the determination of the array, integration, and the net processing gains. Analytical comparisons between conventional and synthetic aperture arrays formed with either the same number of hydrophones or with the same effective length but a different number of hydrophones, show that, when the spatial processing gain exceeds the loss in integration gain, then the use of synthetic apertures is advantageous. [Work performed while at NORDA.]

3:20

LL8. Performance of sinusoidally deformed planar arrays. James H. Leclere, Donald R. Del Balzo, Deanna M. Caveny (NORDA, Code 244, NSTL, MS 39529), George E. Ioup (University of New Orleans, New Orleans, LA 70148 and NORDA, NSTL, MS 39529), Jeffrey L. Becklehimer, and Donald A. Murphy (NORDA, NSTL, MS 39529)

The investigations of Caveny *et al.* [J. Acoust. Soc. Am. Suppl. 1 **80**, S26 (1986)] are extended for the cases of horizontally and vertically towed planar arrays. Horizontal and vertical arrays of three lines (64 hydrophones each) and nine lines (28 phones each) are included. In addition, a nine-line array with five horizontal and five vertical lines (with the middle line of each in common) is also examined. Horizontal sinusoidal deformations of one-half cycle (and in some cases two and three half-cycles) are applied to each line. Complex pressure fields are modeled for various source directions for each array using the code BEAMSTATPAK of Collier (private communication). Beamforming is then done with the known array configuration and with the assumption that the array is planar (for selected cases). Degradations resulting from assuming planarity and the ability to remove left/right ambiguity are summarized as a function of the source location and the amount of array deformation in terms of reduced gain and angular resolution. Array performance is also examined in the presence of a realistic vertical noise distribution. [Research sponsored by NUSC.]

THURSDAY PM

3:35

LL9. Sidelobe suppression in correlated multipath estimates. Peter C. Mignerey (U.S. Naval Research Laboratory, Code 5122, Washington, DC 20375-5000)

Some environments cause a propagating signal to split along several different paths. When such multipath propagation occurs, the covariance among signals traveling along rays emanating from a common source is expected to be larger than the covariance between signals generated by independent sources. An estimate of the covariance between signals arriving from two different directions is shown to be a bilinear form. The ability of the bilinear form to distinguish a correlated arrival from an independent source is studied using a simulated acoustic field. Peaks associated with the correlated pair of arrivals are 24 dB above an uncorrelated background, while no similar peaks occur for the independent source. The 8-dB sidelobes are found to be a problem. An adaptive set of filter vectors, obtained from the classical minimum variance problem, produces a bilinear form which is found to minimize sidelobe interference to 2 dB. However, the peaks indicating a correlated arrival are reduced to 18-dB output above the uncorrelated background. The constrained minimum variance estimate of the covariance between two arrivals of a multipath is apparently effective in suppressing sidelobe interference at the cost of a lower main peak. [Work supported by ONR.]

3:50

LL10. The generation of propeller modulation sounds for sonar trainers. Gary McArthur, Charles Schmid, Mike Warnecke (Honeywell Engineering Services Center, 1050 N.E. Hostmark Street, Poulsbo, WA 98370), and Les Atlas (Department of Electrical Engineering, University of Washington, Seattle, WA 98195)

Sounds of propeller modulation were synthesized using linear predictive coding (LPC) techniques resident in digital signal processing hardware for the purpose of training submarine sonar operators. The combination of LPC techniques and digital sound generation produces realistic propeller modulation sounds which can be conveniently varied by the instructor to characterize the type, speed, and depth of surface and underwater vehicles. The synthesized sounds were generated with digital LPC (all-pole) filters similar to those commonly used in speech synthesizers. The coefficients to drive the LPC filters were calculated from spectral models which were formulated from the cavitation index, K_{TIP}. The cavitation index was calculated based on the instructor's commands of the vehicle type, speed, and depth. Three different LPC filter designs were implemented to evaluate the computational efficiency and sound realism of each approach. The first design was a standard 12th-order LPC filter which was not capable of changing the spectrum over short time periods. The second design was a low order LPC filter with coefficients which changed quickly with time. The last design was a baseband LPC filter modulated by a time-varying sinusoidal carrier. Various propeller modulation sounds will be played to demonstrate the quality and flexibility of these synthesizer techniques for training sonar operators. [Work performed in support of NUWES, Keyport, WA.]

4:05

LL11. Low-frequency ambient noise near Bermuda: Filling in the notch. William A. Von Winkle, David G. Browning, Raymond J. Christian, and A. Donn Cobb (Naval Underwater Systems Center, New London Laboratory, New London, CT 06320)

Ambient noise models which assume no significant wind-generated noise mechanism in the 5- to 100-Hz frequency range predict a notch in noise levels, between shipping and microseismic noise, at approximately 10 Hz. Data are presented for the frequency range 3–200 Hz that show that this notch fills in at increasing wind speeds and is entirely eliminated above 35 kn for a North Atlantic Ocean location near Bermuda. These results are consistent with the low-frequency wind-generated noise mechanism proposed by Isakovich and Kur'yanov [Sov. Phys. Acoust. **16**, 49 (1970)]. [Work supported by NUSC.]

4:20

LL12. Species and size-class discrimination of selected fish using broadband sonar. Paul A. Skvorc, II (Alaska Department of Fish and Game, Comm. Fish Div., P.O. Box 3-2000, Juneau, AK 99802)

A sonar system was designed and built that could transmit and receive a broadband signal. This transmission takes the form of a chirp, ranging in frequency from 50–150 kHz. Four species of fish represented by ten size classes were used in a laboratory environment as targets. An acoustic signature was developed for each of the species and each of the size classes within each species. Pattern analysis techniques were then employed to test unknowns against these signatures. The system as a whole was capable of 95% accuracy in both species and size class discrimination. (Those fish misclassified to specie were considered to be misclassified to size irrespective of the relative sizes of the signature and unknown.) Size-class accuracy is approximately $\pm 10\%$ of the fork length of the fish. Hydroacoustic and pattern recognition techniques are presented.

Joint Meeting of Accredited Standards Committees S3 and S1

The activities of S3 will be discussed first, proceeding to matters of interest to both S3 and S1, and concluding with S1 activities.

Meeting of Accredited Standards Committee S3 on Bioacoustics

L. A. Wilber, Chairman S3
422 Skokie Boulevard, Wilmette, Illinois 60091

Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years.

Meeting of Accredited Standards Committee S1 on Acoustics

D. Johnson, Chairman S1
Larson-Davis Laboratories, 280 South Main, Pleasant Grove, Utah 84062

Standards Committee S1 on Acoustics. Working group chairs will report on their progress in the preparation of standards, methods of measurement and testing, and terminology in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound level meter specifications. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, for which S1 and S3 serve as the U.S. Technical Advisory Groups, will be discussed, with a report on the meeting of ISO/TC 43, held from 4–7 May 1987 in Copenhagen, Denmark.

Session MM. Engineering Acoustics II: Transducers and Sampling

John C. Baumhauer, Jr., Chairman
AT&T Consumer Products Laboratories, P. O. Box 1008, Indianapolis, Indiana 46206

Chairman's Introduction—8:30

Contributed Papers

8:35

MM1. Nonuniformly poled piezoelectric transducers. Dov Hazony (Electrical Engineering and Applied Physics, Case Western Reserve University, Cleveland, OH 44106)

A transducer model similar to Mason's model [W. P. Mason, *Electromechanical Transducers and Wave Filters* (Van Nostrand, New York, 1948)] is presented and discussed. Many examples will be shown including the case when the mechanical response of the transducer to an electrical impulse is also an impulse. The presentation is based on a forthcoming paper [D. Hazony, J. Acoust. Soc. Am. (to be published)]. Corroborative experimental results will be given. [Work supported by Gould, Inc., and Tecsonics, Inc.]

8:50

MM2. Determination of the mechanical limits of piezoelectric transducers using the finite element method. B. Dubus, J. C. Debus, J. N. Decarpigny (Institut Supérieur d'Electronique du Nord, 41 Boulevard Vauban, 59046 Lille Cedex, France), D. Morel (Sinaptec, 41 Boulevard Vauban, 59000 Lille, France), and D. Boucher (Groupe d'Etude et de Recherche de Détection Sous-Marine, Le Brusc, 83140 Six Fours les Plages, France)

The mechanical limit of ultrasonic transducers is associated with the fracture, nonlinearities, and fatigue strength of the different materials and thus is directly related to the distribution of stresses over the structure. The static case corresponds generally to the design of depth transducers and prestressed ceramic stacks. The dynamic case concerns high-power sonar transducers, as well as macrosonics. For several transducers, the stress field has been computed with the help of a finite element modeling, using the ATILA code [J. N. Decarpigny *et al.*, J. Acoust. Soc. Am. **78**, 1499 (1985)]. This paper describes such an analysis in the case of an axisymmetrical length expander transducer and a flexural shell sonar transducer [B. Hamonic *et al.*, J. Acoust. Soc. Am. Suppl. 1 **80**, S26 (1986)]. The static stresses distribution is found when a mechanical bias is applied and maximum dynamic stress zones are identified at several frequencies, using a classical fatigue criterion. Measurements with strain gauges display a satisfactory agreement with computed results.

9:05

MM3. Electroacoustic characteristics of a 1-3 composite transducer. William Thompson, Jr. (Applied Research Laboratory and Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

A 1-3 composite transducer was constructed by dicing a solid square block of PZT-4 to create a large number of thin parallel pillars; the cuts were then backfilled with epoxy. The device was calibrated as both source and receiver and its response was compared with that of the same size solid block. Over the frequency interval 10 to 300 kHz, the transmitting voltage response of the 1-3 composite is comparable in level to that of the solid block, except in the neighborhood of the length resonance of the solid block; moreover, it is a smoother curve. The open circuit receiving response of the composite is some 10 to 20 dB lower over this frequency interval that is attributed to the relatively large capacitance of the cables that were used. Directivity patterns of the 1-3 composite are better than those of the solid block and the availability of multiple outputs makes it possible to implement both amplitude shading for pattern control, and phasing or time delays for beamtilting; both of these features were demonstrated. [Work supported by ONR.]

9:20

MM4. Analysis of multilayered wideband transducers. Hajime Fujita (Mechanical Engineering Research Laboratory, Hitachi, Ltd., 502 Kandatsu, Tsuchiura, Ibaraki 300, Japan) and Kiyoshi Koyano (Totsuka Works, Hitachi, Ltd., Japan)

Single piezoelectric material used as a transducer element usually gives a high Q resonance characteristic due to high acoustic impedance relative to that of water. Epoxy resins or similar materials are used as impedance matching layers in order to obtain a low Q transducer. Although analysis of a composite structure element is traditionally made with an electromechanical equivalent circuit, numerical analysis is increasingly in use. In this paper, a simple and useful design method for a multilayered transducer utilizing both an equivalent circuit and a finite element method is described. Transducers with a half-wavelength piezoelectric material with one or two quarter wavelength matching layers are analyzed for transient response with pulsed sine wave input as well as for steady-state response, and vibration, stress, and sound radiation characteristics are obtained. The results indicate that the best matching impedance is different from the value usually suggested [B. V. Smith and B. K. Gazey, IEE Proc. **113**, 285–297 (1984)].

9:35

MM5. Ultrasonic microprobe hydrophones. Mark B. Moffett, James M. Powers, and William L. Clay (Naval Underwater Systems Center, New London Laboratory, New London, CT 06320-5594)

Miniature, bilaminar ultrasonic hydrophones have been constructed with elements of piezoelectric lead titanate ($PbTiO_3$), lead metaniobate ($PbNb_2O_6$), and poly(vinylidene fluoride) (PVDF). The overall shape of the bilaminar element was cubical, with dimensions of about 0.8 mm for the $PbTiO_3$ and $PbNb_2O_6$ probes and 1.3 mm for the PVDF probe. When used with a 19-dB-gain, 100-pF-input capacitance preamplifier, the sensitivities ranged from -229 dB//1V/μPa for the $PbTiO_3$ to -236 dB//1V/μPa for the PVDF probe. The resonance frequency of the $PbTiO_3$ and $PbNb_2O_6$ probes, about 0.9 MHz, was lower than expected; this is probably due to the relatively high percentage of epoxy used in constructing the elements. If one defines the useful frequency range as that for which the response is flat within ± 1 dB and the beam pattern is omnidirectional within ± 1 dB, the useful upper frequency limit extends to 100 kHz for the (larger) PVDF probes and to 200 kHz for the $PbTiO_3$ and $PbNb_3O_6$

microprobes. These upper frequency limits correspond to element dimensions about one-tenth wavelength in water. [Work supported by NUSC's Independent Exploratory Development Program (funded by DNL) and by the Transducer Technology Project (managed by NRL/USRD and funded by ONT).]

9:50

MM6. Analysis of a magnetostrictive Tonpilz transducer. F. Claeyssen, D. Boucher, and C. Pohlenz (GERDSM, DCAN Toulon, DCN, Le Brusc, 83140 Six Fours les Plages, France)

A rare-earth iron Tonpilz transducer has been developed in a compact size, low-frequency, high-power design. The main reason for this first transducer choice is that its behavior is rather well understood, the rods being used on their longitudinal mode. The active elements of the Tonpilz are four rods of random oriented TbO.3DyO.7 Fel.8 alloy. They are magnetically coupled through a laminated flux return path. Force-cooled coils provide the dynamic magnetic field as well as the polarizing bias. Data on electrical impedance, transmitting current response, transmitting voltage response, effective coupling coefficient, efficiency, and linearity versus the bias are presented. The in-air and in-water measurements are compared with the results predicted by a plane wave model taking into account leakage flux, reluctance of the return path, eddy current losses in the rods and in the return path, and mechanical damping. The material constants of the magnetostrictive alloy used in the model are measured independently with a low-frequency, two-rods device under high-power conditions. New developments in modeling and technology design will be presented in a subsequent paper.

10:05

MM7. Bandwidth limitations of the mechanical impedance of single degree of freedom electromechanical vibrators. Michael P. Johnson (Acoustics and Sensor Systems Group, Gould, Inc., Ocean Systems Division, 18901 Euclid Avenue, Cleveland, OH 44117)

All mechanically resonant transducer devices exhibit a minimum in their mechanical impedance at their fundamental resonance. This minimum in impedance can result in deterioration of electroacoustic performance with possibly disastrous results if left unchecked. The standard technique used to prevent this is attributed to Carson [J. Acoust. Soc. Am. **34**, 1191–1196 (1962)] and essentially provides an electrically induced antiresonance near the mechanical resonance frequency. This technique results in two minima, one residing on each side of the previous minimum. The relative locations of these minima is dependent only on effective electromechanical coupling coefficient and the ratio of the electrical antiresonance to mechanical resonance frequencies. The bandwidth between these minima appears to have a practical limit of approximately an octave.

10:20

MM8. Hydrophonic probe for underwater intensity measurements. S. Pauzin and D. Biron (ONERA/CERT/DERMES, 2 Avenue Edouard Belin-B.P. 4025, 31055 Toulouse Cedex, France)

Two hydrophones are mounted side by side with a separation distance of 20 mm for the frequency range 1–10 kHz. In an anechoic chamber, the probe is validated with a microphonic intensity probe for acoustic power measurements of a reference sound source. The results are compared and agree very well. Directional characteristics of the probe are carried out. Then, an experimental setup is developed in a small cavity ($2\times1\times0.8$ m^3) filled with water. A hydrophone as projector is put in the middle of the tank. The characterization of this test facility is done in terms of the time response between the different boundaries and the receiver. The sound power measurements obtained by an hydrophone associated with a gating system to simulate free-field and those obtained by the hydrophonic probe are compared. Sound source localizations are also investigated. The results are in good agreement in spite of the small bounded water tank. This technique seems to be very interesting to study material characteristics like transmission or absorption and source localization.

10:35

MM9. Electroacoustic mathematical model of a capacitive solid-state microphone structure. John C. Baumhauer, Jr. (AT&T Consumer Products Laboratories, 6612 East 75th Street, Indianapolis, IN 46250)

A *semiconductor microphone* structure is described in terms of the optimization of its micromachining parameters. Microscaling of dimensions is subject to a minimum signal-to-electrical noise ratio requirement, with an objective of small silicon diaphragm area. Side constraints include electrostatic stability, acceptable frequency response, and low-condenser bias voltage. Relations on electrostatic biasing and acoustic sensitivity are presented. A lumped parameter model of the silicon diaphragm was justified for the equivalent circuit. An electrical noise model is described. Scaling to a submicron air film thickness was required to meet the above criteria. A novel (patented) acoustic hole array through the backplate overcomes the resulting (excessive) air squeeze-film damping (i.e., resistance). The array of 2000 holes, nevertheless, consumes a small fraction of the capacitive area. Analyses for mechanical vibration pickup sensitivity and perforated backplate mechanical stability are briefly discussed.

10:50

MM10. The analysis of an optical fiber microphone design. Andong Hu, Frank W. Cuomo (Department of Physics, University of Rhode Island, Kingston, RI 02881), and Allan J. Zuckerwar (NASA Langley Research Center, MS 238, Hampton, VA 23665)

The theoretical response of a condenser microphone [A. J. Zuckerwar, J. Acoust. Soc. Am. **64**, 1278–1285 (1978)] and an electret microphone [R. Zahn, Acoustica **57**, 200–204 (1985)] has been developed based on the mean displacement of a stretched membrane. Recent developments have produced an optical fiber pressure sensor [F. W. Cuomo, R. S. Kidwell, and A. Hu, SPIE **661**, 234–239 (1986)], which utilizes a stretched metallized Mylar film with an optical fiber probe acting at its center. This paper derives the theoretical expressions representing the peak displacement at that point. In addition, the effects of damping due to the backchamber are investigated to show the variations in amplitude and frequency response due to the different design approaches. It is shown that equivalent expressions for the peak displacement can be obtained exactly as well as by the lumped elements approach. Comparisons are made between the results previously obtained for a 1-in.-diam B&K microphone and the present analysis. [Work supported by NASA.]

11:05

MM11. Fiber optic sensor for measurement of pressure fluctuations in a turbulent boundary layer. Part I: Construction and calibration. Frank W. Cuomo, Robert S. Kidwell, and Andong Hu (Department of Physics, University of Rhode Island, Kingston, RI 02881)

The measurement of mean velocities, forces, pressures, and shear stresses can provide useful information about transition, separation, and turbulent friction in the study of fluid flow. In most cases, previous sensors have lacked some of the features necessary to satisfactorily establish the flow properties. This paper discusses the development of an optical fiber pressure sensor whose active dimensions, sensitivity, and frequency response have shown to improve the measuring capabilities presently available. The device is designed for flush mounting in a suitable aluminum plate installed in the test section of a wind tunnel. Details of the construction will be discussed and data, related to the dc and ac calibration methods used, will be presented. [Work supported by NASA and NUSC.]

11:20

MM12. Fiber optic sensor for measurement of pressure fluctuations in a turbulent boundary layer. Part II: Preliminary test results. Allan J. Zuckerwar, Ralph D. Watson (NASA Langley Research Center, Mail Stop 238, Hampton, VA 23665), and Frank W. Cuomo (Department of Physics, University of Rhode Island, Kingston, RI 02881)

The fiber-optic sensor described by Cuomo *et al.* (Part I) was used to measure the pressure fluctuations of a turbulent boundary layer in the 7-$\times$11-in. low-speed wind tunnel at Langley Research Center. The sensor was flush mounted in the bottom plate of the test section at a distance of 30 in. from a trip wire. The free-stream flow speed was varied from 0–140 ft/s in five intervals. At each flow condition the static pressure in a plenum chamber surrounding the test section was adjusted to match the pressure inside to eliminate inflow or outflow. At the same time, the position of one wall was adjusted to provide negligible longitudinal pressure gradients due to the growth of the boundary layer at the tunnel wall. The nondimensionalized power spectral density compares favorably with other measurements reported in the literature. [Work supported by NASA.]

11:35

MM13. On multichannel sampling using only past samples. John L. Brown, Jr. (Department of Electrical Engineering, The Pennsylvania State University, University Park, PA 16802)

A deterministic signal bandlimited to the normalized frequency interval $-\pi < \omega < \pi$ is completely determined by its sample values taken at the Nyquist rate of one sample per second. It is well known that if the signal is processed in parallel by m "independent" linear time-invariant filters (channels), then the signal may be recovered from samples taken at the m samples per second over *all* time. Here, the more realistic problem of estimating $x(t)$ at a generic time t from the multichannel outputs sampled at only past sampling points $\{t - nT\}$ is considered. It is shown that if the m channel outputs are all sampled at the same rate of $1/T$ samples per second, where $0 < T < m$, then $x(t)$ is uniquely determined from the channel output samples taken at the past times $\{t - nT\}$. Moreover, the condition required of the channel transfer functions to assure this result is the same as that required when the samples are taken over the doubly infinite time interval.

FRIDAY MORNING, 15 MAY 1987 REGENCY BALLROOM C & D, 8:30 TO 11:50 A.M.

Session NN. Musical Acoustics V and Psychological and Physiological Acoustics IX: Rhythm and Timing

Punita Singh, Chairman
Central Institute for the Deaf, 818 South Euclid, St. Louis, Missouri 63110

Chairman's Introduction—8:30

Invited Papers

8:35

NN1. Timing in auditory perception. Ira J. Hirsh (Washington University and Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Recognition of sequential patterns of sound is basic to speech recognition, music perception, and identification of environmental sounds from birdsong to thunderclaps. While we define these sound patterns as sequences of brief sounds, the usual acoustical description has principally been the spectrum. For some 30 years, work has been reported on fusion and successivity, duration discrimination, perceived order, and the influence of temporal variables on more traditional psychoacoustic dimensions. At Central Institute, with colleagues Divenyi, Lauter, Grant, Singh, and Monahan, our emphasis has been on listeners' sensitivity to changes in order and timing, and influence of nontemporal features on such perception. Here, some experiments are described that address interval perception in a series. In an isochronous series of three, six, or ten tones with intertone intervals of 200 ms or more, listeners can detect a temporal offset or "jitter" of between 5% and 10%, irrespective of the position of offset. For shorter intervals, performance is worse and there is a strong effect of position in the series, with offsets made late in the pattern being more easily discriminated than those made earlier. When a single pitch change is made in one tone in a series of six, detection of an offset is impaired when it coincides with the different pitch. Phrased patterns (do–re–mi–do–re–mi) on the other hand, have not shown such a strong position effect. It is recognized that temporal patterns influence, and are influenced by stimulus dimensions as well as by dimensions learned from formal systems like the rules of language and music. Our search continues to focus on an in-between perceptual territory, which lies beyond sensory processes, but yet is less dependent on rule learning. [Work supported by NINCDS and AFOSR.]

9:05

NN2. Timing in music: Performance and experience. Alf Gabrielsson (Department of Psychology, Uppsala University, Box 227, S-75104 Uppsala, Sweden)

The only variable over which a music performer has practically complete control, regardless of which instrument he uses, is the duration of the sound events, as well as of "nonsound" events. Timing may therefore be considered as the most important tool available to the performer. Generally seen, timing can be related to

four different phenomena: tempo, different classes of durations, articulation, and deviations from mechanical regularity. The musical score provides only limited information about all these aspects. Empirical studies of music performances reveal that the variations in these variables are much larger and much more frequently occurring than what is usually realized. They are used in order to influence the listener's experience of the music with regard to the perceived structure as well as regarding the motional–emotional qualities of the music. The systematic investigation of these questions is only in its initial stage and faces many difficulties, not the least concerning the proper ways of identifying and measuring the suggested experiential variables. Examples will be given from ongoing research.

9:35

NN3. Determinants of experienced tension in timbre sequences. Dirk-Jan Povel (Psychological Laboratory, University of Nijmegen, P. O. Box 9104, 6500 HE Nijmegen, The Netherlands)

In listening to rhythms, subjects may experience a sensation of tension. It is assumed that this important psychological attribute is directly related to the process of perceiving a rhythm. In this study, tension judgments ascribed to repetitively presented binary isochronic sequences (onset-interval 130 ms) are examined. One "period" of a sample sequence can be notated as X X O X X O X O in which X and O represent two timbres produced by different drums. Subjects were asked to rate the experienced tension of 16 sequences on a five-point scale. Responses show consistent differences in judged tension. Tension judgments are explained by supposing that the sequences are perceived as composed of two separate sequences formed by the X and O events, respectively [Garner, *The Processing of Information and Structure* (Erlbaum, Potomac, MD, 1974)], each contributing its own tension to the compound tension sensation. The tension evoked by the separate sequences is explained by the displaced beat hypothesis proposed by Povel ["Time, Rhythms and Tension: in Search of the Determinants of Rhythmicity," in *Time, Mind and Behavior*, edited by J. Michon and J. Jackson (Springer, Berlin, 1985)].

10:05

NN4. The perception of metrical structure: Experimental evidence and a new model. Christopher Lee (Laboratory of Experimental Psychology, University of Sussex, Brighton BN1 9QG, England)

This paper presents experimental results which are in conflict with existing theories of metrical interpretation [H. C. Longuet-Higgins and C. S. Lee, Perception **11**, 115–128 (1982); H. C. Longuet-Higgins and C. S. Lee, Music Perception **1**, 424–441 (1984); F. Lerdahl and R. Jackendoff, *A Generative Theory of Tonal Music* (MIT, Cambridge, MA, 1983); D. J. Povel and P. Essens, Music Perception **2**, 411–440 (1985)] and a new model, based on the algorithm in Longuet-Higgins and Lee (1982), which accounts for the main features of these results. Musically sophisticated subjects listened to sequences of identical tones in various rhythmic arrangements and transcribed them using standard musical notation. The results disconfirmed key predictions of all four theories, in that subjects displayed a general reluctance to choose interpretations containing an upbeat. However, the results did show that subjects were more prepared to avoid syncopated interpretations than those in which long notes occurred on weak beats. These tendencies are accounted for in the model by the assumption that the listener initially takes the first note of a sequence as the downbeat and is reluctant to abandon this (or later) hypotheses. But the occurrence of syncopations constitutes strong counterevidence and the occurrence of long notes on weak beats weak counterevidence, which can overcome the listener's conservatism. [Work supported by Nuffield Foundation, Royal Society and SERC.]

10:35

NN5. Rhythm perception and the perception of periodicity. Gerald J. Balzano (Department of Music, University of California at San Diego, La Jolla, CA 92093)

To hear something repeating is to hear *something* repeating, and this something must have a beginning and an end. Just where beginnings and ends are perceived in periodic duration patterns is the problem in rhythm perception with which the present studies are concerned. More specifically, how does the metrical framework generated by a periodic duration pattern map into the elements of the pattern? Where is the "downbeat" perceived to occur, and how is the time period initiated by that downbeat organized into subperiods? These questions were addressed using two distinct methodologies. In one, listeners from an introductory musical literacy course transcribed simple patterns into musical staff notation. In the other, listeners from a wider population responded to similar patterns by tapping along in time with them. Both methods revealed pronounced tendencies to hear patterns in particular ways. Some candidates for rhythmical "laws of organization" that account for these tendencies are offered, along with a commentary on the relationship between the two methods of measuring rhythmic interpretations.

Contributed Papers

11:05

NN6. The influence of temporal contour and interval information on the discrimination of rhythmic patterns. Caroline B. Monahan (Central Institute for the Deaf, 818 S. Euclid Avenue, St. Louis, MO 63110) and Edward C. Carterette (Department of Psychology, University of California, Los Angeles, CA 90024)

Montone rhythmic patterns were constructed, each totaling 16 beats apportioned among 6 notes; we describe the rhythms as ordered sequences of beat interval values of 6 notes, e.g., 313144. Musicians judged similarity of standard-comparison pattern pairs. On target trials, the comparison had the same rhythm as the standard; on related trials, the comparison had the standard's temporal contour, that is, the same series of longs, shorts, and sames without regard to interval size (as in the pair 313144--315133); on lure trials, the comparison had a different contour and intervals. Twenty-four musicians were randomly assigned to one of three listening conditions that differed in trial types presented: (A) targets versus lures; (B) targest versus relateds, or (C) relateds versus lures. Two other factors were tempo and metrical simplicity (i.e., the extent to which groups of intervals formed simple submultiples of the 16 beats). Discrimination was better (1) in condition A than in B or C, although performance in both B and C was still above chance, demonstrating the importance of interval and contour information, respectively; (2) when the comparison tempo was the same as the standard's; (3) with metrically simpler standards. Tempo interacted with condition: discrimination of comparison patterns that were faster or slower was poorer in B and C than in A.

11:20

NN7. Expressive microstructure in music: A first assessment of "composers' pulses." Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

According to a provocative theory set forth by Manfred Clynes [most recently in Cognition and Communication **19**, No. 2 (1986)], there are composer-specific patterns of (unnotated) musical microstructure that, when discovered and realized by a performer, help to give the music its characteristic expressive quality. Clynes, relying on his own judgment as an experienced musician, has derived such "pulses" for several famous composers by imposing time-amplitude warping patterns on computer-sythesized classical music. To conduct a formal perceptual asessment of four such pulses, two sets of piano pieces by Beethoven, Haydn, Mozart, and Schubert, one in triple and the other in quadruple meter, each synthesized with each composer's pulse plus a "neutral" version, were obtained from Clynes and presented in random order to listeners of varying musical sophistication for preference judgments. The results show reliable changes in listeners' pulse preferences across different composers' pieces, which supports one essential prerequisite of Clynes' theory. However, there were some significant deviations from the predicted perference patterns. Possible causes will be discussed. [Work supported by NIH-BRSG.]

11:35

NN8. Effects of interpretation on timing in piano performance. Caroline Palmer (Psychology Department, Uris Hall, Cornell University, Ithaca, NY 14853)

Three timing methods in piano performance were described in a previous report [C. Palmer, J. Acoust. Soc. Am. Suppl. 1 **79**, S75 (1986)]: onset asynchronies, rubato patterns, and legato and staccato patterns. The present study examined the relation of these timing methods to performers' intended interpretations of an excerpt. Notated interpretations included indication of primary melody, phrasing, dynamics, and tempo changes. Performances were recorded on a computer-monitored (MIDI interface), velocity-sensitive, weighted keyboard. The performances showed consistent temporal patterns, directly related to the specified interpretations. Onset asynchronies occurred within chords, such that the primary melody (as notated by each performer) preceded the other voices. Asynchronies were generally largest on the first beat of each measure, marking the excerpt's metrical structure. Rubato patterns (deviations in tempo from mechanical regularity) showed larger changes between, than within performers' notated phrases. Legato and staccato patterns within phrases were accurately predicted by a combination of the durations of successive notes in the musical score, but only when performers' intended phrasings were taken into account. Each of these timing patterns decreased in degree when the pianists were asked to play unmusically. [Work supported by NSF and NIMH.]

FRIDAY MORNING, 15 MAY 1987 REGENCY BALLROOM A & B, 8:30 TO 11:45 A.M.

Session OO. Speech Communication IX: Speech Recognition

George D. Allen, Chairman

Department of Audiology and Speech Sciences, Purdue University, West Lafayette, Indiana 47907

Chairman's Introduction—8:30

Contributed Papers

8:35

OO1. An acoustic-phonetic data base. William M. Fisher (Texas Instruments, Inc., Dallas, TX 75266), Victor Zue (Massachusetts Institute of Technology, Cambridge, MA 02139), Jared Bernstein (SRI International, Menlo Park, CA 94025), and David S. Pallett (National Bureau of Standards, Gaithersburg, MD 20899)

DARPA has sponsored the design and collection of a large speech data base. Six hundred and thirty speakers read ten sentences each. Two sentences were constant for all speakers; the remaining eight sentences were selected from a set of 450 designed at MIT and 1890 selected at TI from text sources. The set of sentences is phonetically rich, balanced, and deep. Although all recordings were made in Dallas, we sampled as many varieties of American English as possible. Selection of volunteer speakers was based on their childhood locality to give a balanced representation of geographical origins. The subject population is adult; 70% male; young (63% in their twenties); well educated (78% with bachelor's degree); and predominantly white (96%). Recordings were made in a noise-reducing sound booth using a Sennheiser headset microphone and digitized at 20 kHz. A natural reading style was encouraged. The recordings are complete, and time-registered phonetic transcriptions are being added to the 6300 speech files at MIT. A version of the complete data base (16-kHz sample rate, with acoustic-phonetic transcriptions—approximately 50 megabytes of data) will be made available to researchers through the National Bureau of Standards. [Work supported by DARPA.]

8:47

OO2. Phonetic labeling and acoustic correlates for building Japanese speech data base. Yoshinori Sagisaka, Shigeru Katagiri, and Kazuya Takeda (Advanced Telecommunications Research Institute International, Osaka, Japan)

Fine description of a large amount of speech signals is indispensable to acquire effective acoustic-phonetic rules in speech synthesis and recognition. To build a finely labeled speech data base, manual labeling is carried out using digital sound spectrograms and additional acoustic parameters that reflect power and spectral characteristics. Through labeler training, labeling items are modified several times to decrease the deviation of segment boundaries and to hasten labeling speed. As a result, not only the usual phonemic categories, but also finer phonetic events (e.g., closure, burst, and aspiration for plosive consonants) are labeled. Moreover, multiple segment boundaries (e.g., boundaries between vowels and following fricative consonants) and inseparable portions (e.g., aspiration followed by a devocalized vowel) are specially marked. To ensure labeling quality, reliability tests and error analysis are carried out. Labeling criteria using these results will be used for a large scale data base construction.

8:59

OO3. Evaluation of ASR front ends in speaker-dependent and speaker-independent recognition. Jean-Claude Junqua (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

This paper extends previous experiments of Tsuga and Hermansky [J. Acoust. Soc. Am. Suppl. 1 **80**, S18 (1986)]. Those experiments evaluated the effect of spectral model order, in automatic speech recognition (ASR), using a small alpha-numeric data base. PLP (perceptually based linear predictive) and LP (linear predictive) analyses were compared, using a cepstral and RPS (root power sums) metric. Those experiments dealing with a bigger data base were validated (104 words, ten speakers). PLP RPS front end is compared with about ten other ASR front ends (LP cepstrum, LP RPS, critical band,...). Experiments were run at various different analysis model orders. Results of speaker-dependent ASR show that the low-dimensional PLP analysis is a good alternative to high-dimensional LP or filter bank analysis. The index-weighted metric improves the recognition accuracy, making the recognition results more uniform across the speakers. The speaker-independent experiments (which use the templates of one male and one female speaker as references) confirm the superior performance of the PLP method for extracting speaker-independent information. Most of the errors are on the consonants; some improvements on the recognition process are being investigated.

9:11

OO4. Speaker-independent vowel classification based on fundamental frequency and formant frequencies. James Hillenbrand and Robert T. Gayvert (RIT Research Corporation, Rochester Institute of Technology, Rochester, NY 14623-3435)

A multivariate distance measure (MVD) was tested on a data base consisting of hand-measured fundamental and formant frequency values from ten English vowels produced by 29 male talkers and 27 female talkers. The primary purpose of the study was to determine what set of parameters produced the best classification performance. Results included: (1) classification accuracies as high as 90% were obtained using exclusively internal information (i.e., only acoustic measurements of the unknown token); (2) classification accuracies as high as 95% were obtained when internal information was combined with information describing the talker (e.g., average formant frequencies); (3) parameter sets using absolute formant frequencies performed better than sets using formant ratios or log formant distances; (4) a very small decrement in performance (1.1%) was observed when MVD was tested under conditions of no overlap between the talkers used to train the system and the talkers used to test the system; and (5) MVD did not need to be trained separately on male and female talkers. [Work supported by the Air Force Systems Command, Rome Air Development Center, and the Air Force Office of Scientific Research, Contract No. F30602-85-C-0008.]

9:23

OO5. Predicting stress and syllable boundaries from segmental timing. Sven Anderson, Robert Port (Department of Linguistics, Indiana University, Bloomington, IN 47405), and Daniel Maki (Department of Mathematics, Indiana University, Bloomington, IN 47405)

Recent studies [Reilly and Port, J. Acoust. Soc. Am. Suppl. 1 **78**, S21 (1985)] show that timing measurements can be used to discriminate among items in a small vocabulary. Current efforts question the extent to which timing measurements can be used to augment basic segmental knowledge in a continuous, real-time speech recognition situation. These experiments examine a single template of segments (vowel–strong fricative–stop–vowel) each of which is embedded in different words in 112 different sentences for four English speakers. Syllable and word boundaries occur between each of the four segments. Discriminant analysis is used to predict the stress patterns, possible syllabifications, and word boundaries from timing measurements in the templates of a fourth male speaker after training on three others. Results indicate that within the template a linear combination of simple timing measurements can be used as predictors of lexical stress (95% correct). Word and syllable boundaries are much less reliably predicted (60%–75%) and apparently bear no simple linear relationship to timing measurements. [Research supported by NSF.]

9:35

OO6. Acoustic phonetic representations for continuous speech recognition: Networks versus lattices. Robert A. Brennan and Michael S. Phillips (Department of Computer Science, Carnegie-Mellon University, Pittsburgh, PA 15213)

A lattice has been used to represent the acoustic phonetic hypotheses in the Carnegie Mellon speech recognition system. This lattice was produced by four separate modules. Each module independently located and classified a set of target segments. A network representation is now being used that explicitly defines allowable paths through the phoneme hypotheses. The network has two advantages over the lattice representation. First, acoustic phonetic information can be used to join segment hypotheses. Second, a network allows the use of context-specific phonetic classification. Initially the network has been produced from the lattices using acoustic rules and broad classification information to connect segments, to adjust boundaries between segments, and to fill in gaps between the segments. Context-specific classification is now being developed for phones that are particularly sensitive to context effects. This network representation is being evaluated in the framework of the overall Carnegie-Mellon system. A comparison of the network representation with the lattice representation will also be presented. [Work supported by DARPA.]

9:47

OO7. Learning phonetic features using connectionist networks. Raymond L. Watrous and Lokendra Shastri (Department of Computer and Information Science, University of Pennsylvania, Philadelphia, PA 19104)

A method for learning phonetic features from speech data using connectionist networks is described. A temporal flow model is introduced in which sampled speech data flow through a parallel network from input to output units. The network uses hidden units with recurrent links to capture spectral/temporal characteristics of phonetic features. A supervised learning algorithm is presented which performs gradient descent in weight space using a course approximation of the desired output as an evaluation function. A simple connectionist network with recurrent links was trained on a single instance of the work pair "no" and "go," and successfully learned a discriminatory mechanism. The trained network also correctly discriminated 98% of 25 other tokens of each word by the same speaker. The discriminatory feature was formed without segmentation of the input, and without a direct comparison of the two items. The network formed an internal representation of a single, integrated spectral feature which has a theoretical basis in human acoustic-phonetic perception.

9:59

OO8. Text-independent speaker verification using linear predictive hidden Markov models. Naftali Tishby (Speech Research Department, Rm. 6H-506, AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974-2070)

The application of hidden Markov models to text-independent speaker verification was studied. Linear predictive hidden Markov models have been proved to be an efficient way for statistical modeling of speech signals in all the forms of speech recognition. As was already suggested by Alan Poritz, such models can be used for statistical characterization of the talker himself. In our case, ergodic Markov models with four–seven states, are shown to discriminate between speakers, where the spectral density of the states is characterized by eight order LPC coefficients. A large data base of 100 talkers with about 20 000 isolated digits was used. The states, as well as the optimal model dimensionality, tend to supply enough information to verify the talker with less than 3% error rate, on sufficiently large testing data. The results show, however, that most of the speaker-dependent information is contained in the spectral densities and very little is left in the transition matrix. This may justify the vector quantization approach to the problem.

10:11–10:21

Break

10:21

OO9. Automatic speech recognition in a noisy automotive environment. J. G. Wilpon, L. R. Rabiner, D. DeMarco, and K. L. Shipley (AT&T Bell Laboratories, Office 2C-573 Murray Hill, NJ 07974)

Recently, a great deal of interest has been shown in performing reliable speech recognition in an automotive environment. In order to determine the level of recognition accuracy to be expected in such a hostile environment, a series of experiments was performed. The data base for these experiments was created by having four talkers (two male, two female) speak ten replications of an 11-digit vocabulary, zero–nine and oh (in an isolated format), while driving in an automobile. Five different environmental conditions were studied: (1) car off, fan off; (2) car idling, fan on high; (3) car traveling at 30 mph, fan on high; (4) car traveling at 60 mph, fan on high; and (5) car traveling at 60 mph, fan off. Two different recognition algorithms were tested, namely; (1) standard LPS-based template recognition, using dynamic time warping; and (2) continuous density hidden Markov model recognition, using Viterbi scoring and cepstral analysis. Several different distance measures were also evaluated for each of the recognition sytems. Results from the template-based algorithms showed speaker-dependent recognition accuracies from 96%–99% depending on which condition was used for training. A five-state HMM system yielded recognition accuracies from 93%–98% depending on the training condition.

10:33

OO10. Utilizing phase spectra for emulative speech recognition (ESR). Nathan Cohen (Department of Natural Sciences, Bentley College, Waltham, MA 02254)

A new approach is described to machine speech recognition that incorporates nonlinear spectral interferometry to model the binaural advantage in human speech recognition. This ESR scheme uses interharmonic and interear visibility observables from the spectra of phonemes to provide phoneme identification signatures.In particular, the phase spectra of phonemes are found to be coupled, but not redundant to, the amplitude spectra. Use of amplitude and phase spectra allows a critical resolution of phoneme ambiguities often present when derived from amplitude spectra alone. Interharmonic phase modulation appears to be a secondary means of encoding human speech and is primarily used to discern the speaker's identification and mood.

10:45

OO11. Dynamic frequency warping, the dual of dynamic time warping. Edward P. Neuburg (National Security Agency, Fort Meade, MD 20755)

Comparison of two tokens of the same utterance is central to many automatic speech recognition systems. Matching is usually done in the frequency-time domain; token matching is effectively spectrogram matching. Dynamic time warping (DTW) overcomes, to some extent, the temporal variability of speech tokens; spectrograms are time-aligned by calculating similarity scores between segments of speech, now represented as "columns" of their spectrograms, and applying the mathematical technique called dynamic programming. DTW distorts the time scales of the spectrograms so that identical speech events in the two spectrograms now occur at identical times. Variability in frequency of these events is normally dealt with only by using robust distance measures. There is a better way. After time alignment, frequency variability can be dealt with specifically by doing a dynamic frequency warp (DFW), a process strictly analogous to the DTW. The "rows" of the speech spectrogram, which show the time behavior of the spectral components, play the same role in the DFW as the "columns" do in the DTW. Distances between the rows are calculated and passed to a dynamic program. The resulting DFW produces a distortion of the frequency scales such that identical speech events in the two tokens now occur both at the same time and at the same frequency. Experience with distance measures between rows is limited, but early results are promising: (1) Tokens that have been warped in both directions match better than tokens warped only in time. (2) Redoing the DTW after a DFW results in improved time alignment. (3) Tokens sound more like each other after DFW than before. (4) Pairs of speakers produce consistent frequency warps. Results will be demonstrated, and other applications suggested.

10:57

OO12. On optimizing the innovation codebook of stochastic coder. Daniel Lin (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

In stochastic coding, the speech waveform is represented as a nonstationary Gaussian random process and is reconstructed by filtering an i.i.d. Gaussian source (innovation) with a time-varying linear filter [M. R. Schroeder and B. S. Atal, Proc. ICASSP 85]. The design of optimum block codes for the white Gaussian innovation using a "polar" product code representation is considered. Each codeword in the product codebook consists of a gain and a vector's "orientation" (specified by a set of vectors on the unit hypersphere). The amplitude factors of the codewords are encoded with a Max quantizer. The codeword "orientations" are iteratively generated from an ensemble of random spherical vectors so as to maximize the minimum spherical angle between the orientation vectors. The product codes are jointly searched to determine the optimum Gaussian innovation. This approach is compared with probabilistically generated codes as well as iteratively generated (cluster) codes for performance. Preliminary simulation results have shown an improvement of nearly 1 dB in the average SNR.

11:09

OO13. Evaluating spectral distance measures with reference to human perception. Y. Kane-Esrig (Cornell University, Ithaca, NY 14853), L. A. Streeter, C. Kamm, S. Devlin, and M. Macchi (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

Two important criteria for spectral distance measures in automatic speech recognition are: (1) the variance of distances between different tokens of the same utterance should be small and variances of distances between different utterances large, and (2) the pattern of distances across utterances should correlate with perceived phonetic similarity of the utterances. These criteria were used to evaluate several spectral distance measures, including (1) Euclidean distance between two log formant ratios, (2) LP-residual ("Itakura") distance, (3) Manhattan distance between linearly spaced points on LP spectra, and (4) Manhattan distance between points on "perceptual" spectra (transforming the frequency scale to barks and convolving with an asymmetric filter of critical bandwidth). Distances were computed among synthetic utterances and among one speaker's natural utterances of 11 Dutch vowels produced in isolation. The perceptual similarity data were those reported by Pols, van der Kamp, and Plomp [J. Acoust. Soc. Am. **46**, 458–467 (1969)]. Not surprisingly, Euclidean distances between log formant ratios predicted perceived similarity best, but this measure is of limited practical utility, since formant tracking is problematic. The second best measure for both evaluation criteria was distance between "perceptual" spectra. Distance calculations using these "perceptual" spectra are computationally feasible, produce the desired spread in intervowel distance distribution, and mimic perceived similarity.

11:21

OO14. Evaluation of a speaker-dependent recognition metric as a substitute for human judgments of speech quality. C. S. Watson, D. Kewley-Port, D. Maki, and D. Reed (Departments of Speech and Hearing Sciences and Mathematics, Indiana University, Bloomington, IN 47405)

The Indiana Speech Training Aid project (ISTRA) is evaluating the use of speaker-dependent speech recognition to provide speech-drill feedback for deaf or misarticulating children. The ISTRA systems employ IBM PC's with Interstate Voice Products Vocalink SRB boards. The basic approach is to form templates from the child's best current productions. Feedback in drill sessions is then based on a goodness metric. That metric represents the "match" between 10-ms sampling of a new utterance and the corresponding information for the stored template. The goodness metric has been partially validated as a substitute for feedback judgments provided by a human teacher. In general, the method compares values of the goodness metric to actual ratings by a jury of listeners. For words spoken by either normal speakers who intentionally vary the quality of their pronounciations, or by hearing-impaired speakers, average inter-listener correlations for ratings of quality did not differ from the correlations between the goodness metric and the average listener's judgements. Test–retest reliability is, of course, much higher for computer than for the human listener. [Work supported by NSF.]

11:33

OO15. The Indiana Speech Training Aid (ISTRA): First year report. D. Kewley-Port, C. S. Watson, M. Elbert, and G. DeVane (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

The ISTRA systems [Watson *et al.*, J. Acoust. Soc. Am. Suppl. 1 **81**, Abstract OO14 (1987)] are used to improve the speech of deaf or misarticulating children, by providing unidimensional feedback based on acoustic information distributed over whole words or other utterances. This approach differs from that of many previous aids in which feedback was based on measures of single acoustic dimensions, or others that provided spectrographic displays. Using software developed over the first year of the project, the clinical effectiveness of ISRA drill has been evaluated for several deaf and one normal-hearing, misarticulating child. The research emphasis has not been on comparisons between the progress of training with a teacher and that with the ISTRA system, but rather on demonstrations that ISTRA drill is itself an effective training method. In these studies, listener juries rate the speech of words collected before, during, and after training. Results show significant improvement of the quality of words trained in the ISTRA drill, as well as generalization to nontrained words. These and other results demonstrating the improvement of articulation after drill on the ISTRA systems will be presented. [Research supported by NSF.]

FRIDAY AM

Session PP. Noise V: Noise from Air Flows and Transportation, Community Noise Modeling, and Propagation and Effects

Julia D. Royster, Chairman
Environmental Noise Consultants, Inc., P.O. Box 144, Cary, North Carolina 27511-0144

Contributed Papers

9:00

PP1. Mechanisms of sound-producing instabilities of choked round jets impinging on small and large flat plates. Alan Powell (Department of Mechanical Engineering, University of Houston, Houston, TX 77004)

At the 112th Meeting certain characteristics of the discrete tones emitted when a choked round jet impinges on a normal flat plate were reported [J. Acoust. Soc. Am. Suppl. 1 **80**, S9 (1986)]. These are now shown to be one of two classes, dependent on whether the plate is small or large. If small ($\cong$orifice diameter D), the action occurs when the stand-off shock-wave is in a pressure recovery region of the spatially periodic jet structure, as in a Hartmann resonator, and the frequency overlaps those of a Hartmann resonator, successively excited at high harmonics. If large ($>4D$), feedback to the orifice occurs, with up to seven frequency jumps *not* correlated with the periodic jet structure:

$$f=\frac{c_0}{(h+h'')}\frac{N}{(1+M_{\rm con}^{-1})},\quad 1\lesssim N=\text{integer}\lesssim 8,$$

where $N \cong h/D+3/2$ $(2<N<8)$ for the *principal* frequencies, c_0 is the ambient sound speed, h is the orifice–plate distance, h'' is a constant $\cong 4D/5$, and $M_{\rm con}$ is the jet instability convection Mach number. These are almost independent of jet velocity and plate size, differing significantly in character from the subsonic case.

9:15

PP2. The feedback mechanism of supersonic edge and plate tones. Alan Powell (Department of Mechanical Engineering, University of Houston, Houston, TX 77004)

Intense tones are generated when supersonic (choked) "2-D" jets impinge on edges [Powell, Acoustica **3**, 233–243 (1955)] or on normal flat plates [Krothapalli, AIAA J. **23**, 1910–1915 (1985)] considered herein equivalent to a 180° edge. A feedback mechanism is now proposed: Puffing to alternating sides of the high-impedance midplane, the jet generates essentially monopole radiation into half-or quarter-space, respectively. The resultant sound pressure p' at the nozzle changes the (quasistatic) efflux angle by $p'/\rho_j c_j\ (M_j^2-1)^{1/2}/M_j$ (M_j is the jet Mach number), as for choked jet screech. The induced jet sinuosity is amplified by flow instability until interacting with edge or plate. The phase criterion of I yields the *principal* frequencies:

$$f=\frac{c_0}{h'}\frac{N+p}{(1+\overline{M}_{\rm con}^{-1})},\quad N=\text{integer},$$

where h' is the nozzle to effective source distance, c_0 is the ambient sound speed, $\overline{M}_{\rm con}$ is the averaged instability convection Mach number, and $p=1/4$. Limited experimental data yield $p\cong 0.25$, $N\cong 7$ for edge tones; for plate tones $2<N<5$, and $p=0$ (sometimes, but $\overline{M}_{\rm con}$ values are uncertain!). Again following I, the gain criterion yields sound intensity and lateral jet displacements consistent with experiment.

9:30

PP3. A time-frequency domain representation of sound energy by use of Wigner distribution. Hideo Suzuki, Jun'ichi Kawaura, and Takahiko Ono (Ono Sokki Company, Ltd., 2-4-1 Nishishinjuku Shinjuku-ku, Tokyo 163, Japan)

A time-averaged sound intensity is calculated from the cross spectrum of pressure outputs $p_1(t)$ and $p_2(t)$ of two closely located microphones. A major drawback of this sound-intensity measurement technique is that it is not applicable to a transient sound. This problem may be solved by use of the Wigner distribution that, roughly speaking, represents an energy density in the time-frequency domain when it is applied to proper functions of time or frequency [T. A. C. M. Classen and W. F. G. Mecklenbräuker, Philips J. Res. **35**, 217–250 (1980)]. The cross-Wigner distribution of the pressure $p(t)$ and the particle velocity $u(t)$ (in the direction of interest) is obtained from the cross-Wigner distribution of $p_1(t)$ and $p_2(t)$ such as $\mathrm{Re}[W_{pu}(t,\omega)]=-\mathrm{Im}[W_{p_1p_2}(t,\omega)]/\omega\rho d$, where d is the microphone distance. Results of the application of the cross-Wigner distribution to several transient signals captured by a microphone probe used for the conventional sound-intensity measurement will be reported.

9:45

PP4. Reduction of the noise of a model counterrotation propeller at cruise by reducing the aft propeller diameter. James H. Dittmar (National Aeronautics and Space Administration, Lewis Research Center, Cleveland, OH 44135) and David B. Stang (Sverdrup Technology, Inc., Middleburg Heights, OH 44130)

The forward propeller of a model counterrotation propeller was tested with its original aft propeller and with a reduced diameter aft propeller. Noise reductions with the reduced diameter aft propeller were measured at simulated cruise conditions. Reductions of up to 7.5 dB were measured for the aft propeller blade passing tone, and reductions in the harmonics were also measured. Reductions in the interaction tones were observed as the result of the reduced diameter aft blades no longer interacting with the forward propeller tip vortex. Significant reductions in the total noise at each harmonic were observed. The chief noise reduction at each harmonic came from reduced aft propeller-alone noise, with the interaction tones contributing little to the totals at cruise. Total cruise noise reductions, as much as 3 dB at given angles for the blade passing tone, and as high as 10 dB at some of the harmonics were observed. These reductions would improve the fuselage interior noise levels and represent a cruise noise benefit for using reduced diameter aft propeller blades.

10:00

PP5. The effect of probe inertia on the accuracy of structural-intensity measurements in lightweight, built-up structures. Robert J. Bernhard and John D. Mickol (School of Mechanical Engineering, Ray W. Herrick Laboratory, Purdue University, West Lafayette, IN 47907)

Two accelerometer methods have been proposed for measurement of flexural wave-energy propagation in beams and plates. In this investigation, the influence of the probe inertia was investigated both analytically and experimentally. It was found that for lightweight structures and narrow-band analysis the probe inertia changes the source impedance and thus significantly affects the intensity field throughout the structure. Power flow accountancy, done using contour integrals of intensity, was found to be highly suspect due to these inertial effects. When probe inertia is compensated by putting dummy probes at all measurement locations, the power flow appears to be consistent and stationary. The results indicate the need for either extremely lightweight or noncontacting structural-intensity transducers. In addition, the investigation points out the care which must be taken to make valid structural-intensity measurements. [Work supported by NASA Langley Research Center.]

10:15

PP6. Reduction of wind noise using correlation in outdoor acoustical measurements. John Brunner, Richard Raspet, and Paul Schomer (U. S. Army Construction Engineering Research Laboratory, P. O. Box 4005, Champaign, IL 61820-1305)

Accurate outdoor acoustical measurements are difficult under high wind conditions. Use of two microphones and windscreens may significantly reduce wind noise. The basic technique uses two facts. First, and most important, is that as the separation of the two microphones increases, the turbulence of the wind becomes largely uncorrelated. Second, the turbulence created by placing the microphone in a steady wind is uncorrelated. A cross-multiplication algorithm on the data from the two microphones will show the effectiveness of this process as compared to using just a windscreen.

10:30

PP7. Echoes from hills. Allan D. Pierce and Pei-Tai Chen (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The question is raised as to whether echoes from hills are caused by reflection, scattering, or diffraction. In a typical situation when echoes are heard from a distant hill, it is impossible to construct a geometrical acoustics path that connects source to hill to listener and yet for which the angle of reflection equals the angle of incidence at the hill's surface. If the hill is modeled as a trapezoid, then the echoes are diffracted waves that originate at the baseline (where hillside meets level ground) and at the hilltop edge (where hillside meets hilltop). For hills with a continuously curved surface, the curvature radius is much larger than the wavelength, rigorous wedge diffraction theory is inapplicable and, moreover, the previous formulations of the geometrical theory of diffraction by curved surfaces also seem inapplicable because they account for diffraction effects only on the shadow side. Recent work by Medwin and others also suggests that the validity of physical optics approximations is questionable. In an attempt to understand the general physics of hill echoes, the classic rigorous solution for cw plane-wave incident on a hard parabolic cylinder has been studied. The contour integral with Whittaker functions in the integrand is approximated in the high-frequency limit for remote points back toward the source. [Work supported by NASA Langley Research Center.]

10:45

PP8. Propagation of sound above a curved surface. Alain Berry (Université de Sherbrooke, Sherbrooke, Canada and Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada) and G. A. Daigle (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

There is a useful analogy between flat ground and curved ray paths in the atmosphere, and a curved ground surface above which there is no refraction. Many features of the sound field in the refractive shadow can be studied under controlled conditions indoors over a carefully constructed curved surface. Preliminary measurements, and comparison with simple theory restricted to the ideal cases of rigid and pressure-release boundaries, have been presented at an earlier meeting [J. Acoust. Soc. Am. Suppl. 1 79, S20 (1986)]. The measurements have been completed in the frequency range between 0.3–10 kHz, above a rigid boundary and a surface of finite impedance. Particular attention is given to the region across the shadow boundary. In the shadow, the theory has been extended by removing restrictive approximations and by calculating the creeping wave to higher-order terms. A numerical algorithm allows the extension to the case of a finite impedance. Above the shadow boundary the sound field is calculated using geometrical theory that accounts for reflection from a curved surface. At the shadow boundary both theories typically agree to within 0.5 dB. Good agreement between measured results and theory is obtained in general, except across the shadow boundary where discrepancies between 2 to 5 dB are observed. The theory may also predict a deeper shadow than measured at higher frequencies above the surface of finite impedance.

11:00

PP9. Experimental investigation of the diffraction of sound by a curved surface of finite impedance. Yves H. Berthelot, James A. Kearns, Allan D. Pierce, and Geoffrey L. Main (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Laboratory scale experiments have been conducted to study the diffraction of sound by a single curved ridge on top of a flat table. The sound source is an electric spark that produces an N-wave type transient of about 40-μs duration. Two $\frac{1}{4}$-in. microphones capture the signal at a reference position (before the ridge), and at a field point, respectively. The signals are gated to eliminate undesirable reflections, digitized (250 kHz/channel), low-pass filtered (50-kHz cutoff frequency), and processed by an IMB PC which computes the Fourier transforms of both reference and field pressure waveforms. Regardless of the spark variability, the ratio R of the Fourier transforms is constant to within 0.5 dB for the frequency range of interest (5–40 kHz). Consequently, R represents the insertion loss of the ridge as a function of frequency. Experimental results include, for two different surface impedances (1) the insertion loss on the curved surface, and (2) the diffraction pattern in the transition region between the illuminated and shadow zones (penumbra), measured at a horizontal distance of 0, 0.71, and 2.09 m from the apex of the ridge. Data will be compared with theoretical results presented earlier. [Work supported by NASA.]

11:15

PP10. Effects of site geometry on noise control investigated by a scale-model experiment. Selma Kurra (Department of Architecture, Division of Physical Environment, Istanbul Technical University, Istanbul, Turkey)

The geometrical parameters of a building and building groups were simulated in a scale-model study in which a miniature air-jet source moving along a channel in a semianechoic room was used. The scale factor was 20:1 and the frequency range was chosen between 1.2–20 kHz, corresponding to 63–1000 Hz of a traffic noise spectra. The noise generated by the source pass-bys was recorded at 2.8-s intervals and analyzed with the aid of computer programs to obtain time/level distributions in various points in the site model, and to determine the overall and spectrum levels of the line source, by also taking into account the typical spectrums for light and heavy vehicles, air absorption, and microphone directivity. After the validity tests of the technique, the effect of multiple reflections from buildings of different heights, variation of gap size between blocks, and dimensions of barrier buildings and their location angles were studied. The correlations and regressions between the physical dimensions and the noise levels were demonstrated. The results were confirmed by the earlier results of some theoretical and experimental investigations. The further evaluations yielded some simple but useful relationships to estimate the noise propagation in urban environments, for the designer's use especially in noise mapping, which requires a large number of noise data. [The scale-model study was carried out at Purdue University, R. W. Herrick Laboratories, with the consultation of Dr. M. Crocker (1981–82).]

11:30

PP11. Plans for a before and after study on community response to traffic noise. Truls Gjestland (ELAB, The Norwegian Institute of Technology, N-7034 Trondheim, NTH, Norway) and Sigurd Solberg (KILDE, P. O. Box 229, N-5701 Voss, Norway)

Major changes in the road traffic pattern will be implemented in Oslo in the near future. A multidisciplinary field study has been planned to assess the impact on the community concerning social, medical, and behavioral effects. More than 1000 dwellings will be affected. In the test area new by-passes/tunnels, traffic calming, and traffic management schemes, as well as facade insulation, will be introduced. The plans for describing changes in the noise exposure situation and the community response to these changes will be discussed.

Session QQ. Physical Acoustics VII: Propagation Phenomena

Leon A. Frizzell, Chairman

Department of Electrical and Computer Engineering, University of Illinois, 1406 W. Green Street, Urbana, Illinois 61801

Contributed Papers

9:00

QQ1. A general-purpose parabolic equation model for atmospheric and ocean acoustics. Michael J. White and Kenneth E. Gilbert (National Center for Physical Acoustics, University of Mississippi, University, MS 38677)

A wide-angle parabolic equation model has been developed that is applicable to a variety of propagation problems in the atmosphere and ocean. Linear finite elements are used to discretize the vertical dependence of the acoustic field. This method of discretization enables the model to efficiently handle both small-scale and large-scale vertical variations in complex wavenumber and density. For example, the model can be applied to a turbulent atmosphere and to a randomly layered ocean sediment. The computational grid is terminated at the top and bottom with a complex impedance condition so that a locally reacting surface for atmospheric problems and a pressure-release surface for underwater problems can easily be incorporated. Results from selected applications will be presented to demonstrate the capabilities of the model.

9:15

QQ2. Elastic wave propagation on a multilayer interface between two solids. S. I. Rokhlin (Department of Welding Engineering, Ohio State University, Columbus, OH 43210)

The propagation of waves on a multilayered interface is investigated by matrix method. The solution is obtained as a series expansion with respect to the thickness of the interface layers. In the first approximation the solution is determined by a simple characteristic equation for the velocity of the interface wave, where the interface layer is described by effective elastic properties, depending mainly on shear moduli. The results are compared with the case of thin surface layers when the effective properties depend on both elastic constants. The theoretical results obtained can be used for calculation of the elastic properties of interface layers based on experimental data on surface and interface wave velocity. Such applicability is demonstrated by several experimental examples.

9:30

QQ3. Nonuniqueness of solutions to variationally formulated acoustic radiation problems. Xiao-Feng Wu and Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Determination of surface acoustic pressure when the normal velocity is prescribed over a vibrating body's surface can be formulated in various ways, but for some such formulations the solution is not unique at certain discrete frequencies. The question arises as to whether uniqueness problems are present for variational formulations of this problem. The answer is ordinarily "yes," but proper selection of the variational approach can help to circumvent such difficulties. The variational formulation based on the normal derivative of the Kirchhoff–Helmholtz integral has a unique solution for vibrating disks and platelike bodies. For bodies of finite volume, but for which each surface point is vibrating in phase, calculated total radiated acoustic power is always unique, even though the pressure may not be. Use of the Gerjuoy–Rao–Spruch technique and judicious choice of what physical quantity is stationary under variations leads to a formulation such that the surface pressure is unique for any specified frequency. Similarities with the Burton–Miller techniques are discussed; an appropriate variational formulation provides a method for choosing the weighting factor for the linear combination of the two surface relations (an integral equation and an integrodifferential equation). [Work supported by ONR.]

9:45

QQ4. Acoustic insertion loss from compliant plate gratings. C. George Ku (Martin Marietta Baltimore Aerospace, Baltimore, MD 21220)

The insertion loss from multiple gratings of compliant plate immersed in water is analyzed. The symmetry condition of the geometric configuration provides the convenience of using normal mode solutions of the equation of motion for plates. The structural motion of the compliant plates is coupled with the acoustic pressure of the surrounding water. Therefore, the magnitude of the insertion loss is controlled by the structural resonant modes of the plates. Numerical results are presented for compliant plate gratings with simply supported end conditions and clamped conditions. A brief discussion about the dynamic behavior of the grating is also presented.

10:00

QQ5. Determination of Biot–Stoll pore-size parameter using high-frequency backscattering. Kevin L. Williams, Roger H. Hackman, and D. H. Trivett (Naval Coastal Systems Center, Code 4120, Panama City, FL 32407)

Calculations utilizing a Biot–Stoll model require knowledge of the pore-size parameter of the sediment. This parameter is difficult to obtain even from *in situ* measurements. Recent experiments using a broadband, high-frequency (0.5–2.5 MHz) pulse incident on a water–sediment interface at angles from 15°–45° (relative to the interface) indicate the possibility of determining this parameter remotely. When the spectrum of the backscattered signal is normalized to that of the incident pulse, an enhancement characteristic of the pore-size parameter becomes apparent. The spectrum shows a gradual rise in amplitude with increasing frequency followed by a decline whose rate depends on the physical parameters of the sediment being examined. Typical results will be shown for several sediments, along with a phenomenological model that recaptures the general features of the experimental results. The model displays the underlying physical effects that should be accounted for in a more comprehensive theory. Included in the model are contributions to scattering from both sediment grains and sediment channels. Also included is a "cutoff" of the channel contributions due to the acoustic scattering cross section of the grains exceeding their geometrical cross section. [Work supported by ONR.]

10:15

QQ6. Pore-fluid wave propagation constants in stacks of low-flow resistivity glass beads. James M. Sabatier and Henry E. Bass (Physical Acoustics Research Laboratory, University of Mississippi, University, MS 38677)

Measurements of the attenuation and phase speed of the pore-fluid wave were made in air-filled unconsolidated stacks of glass beads in the frequency range, 1–10 kHz. Glass bead sizes ranged from 1–10 mm. The experimental apparatus consisted of a 0.25-m-diam aluminum tube. A 0.25-m-diam capacitance transducer was operated in a pulse-echo mode at one end of the tube. At the other end, various depths of glass beads could be supported by a rigid plate. Multiple echoes from within 10–30-cm-thick stacks of beads could be observed. The measured propagation constants are compared to those predicted by the rigid porous frame theory [K. Attenborogh, J. Acoust. Soc. Am. **81**, 93–102 (1987)]. [Work supported by WES.]

10:30

QQ7. Investigation of possibility of damage from acoustically coupled seismic waveform from blast and artillery. James M. Sabatier, Henry E. Bass (Physical Acoustics Research Laboratory, University of Mississippi, University, MS 38677), and Richard Raspet (CERL–EN, Box 4005, Champaign, IL 61820)

Noise from artillary fire causes complaints around military installations. Cracked house foundations are often included in these complaints. For the moderate size explosives involved and the minimum distances (>2 km) to homes, it is improbable that directly induced seismic vibrations would have high enough amplitudes to cause this kind of damage. A possibility that has not been previously investigated is that the acoustically induced seismic wave might have a high enough amplitude to damage foundations. Measurements of the acoustic wave and the acoustically induced seismic wave were made for 8-in. artillery shells and 500-lb bombs with peak levels up to 145 dB, and it was found that the induced seismic wave was much smaller than the minimum criteria for damage. In addition, an acoustic-to-seismic steady-state plane-wave model was modified to predict the acoustic coupled seismic wave. Comparison between predicted and measured results will be presented. [Work supported by WES, CERL.]

10:45

QQ8. The use of CHIEF to obtain unique solutions for acoustic radiation using boundary-integral equations. A. F. Seybert and T. K. Rengarajan (Department of Mechanical Engineering, University of Kentucky, Lexington, KY 40506-0046)

This report is concerned with the problem of obtaining a unique solution for radiation problems at characteristic wavenumbers when a boundary-integral equation formulation is used. It is shown that the combined Helmholtz integral equation formulation (CHIEF) method works even when numerous interior points lie exactly on nodal surfaces of a mode corresponding to an eigenfrequency of the related interior problem, when at least one interior point does not lie on a nodal surface. The use of CHIEF is illustrated with numerical radiation experiments for the sphere and the finite cylinder. CHIEF is compared to the Helmholtz gradient formulation (HGF) for circumventing nonuniqueness and is found to yield a more accurate solution, in at least one published example. A procedure is used to indicate the solution error when using integral equation methods, with or without a technique to circumvent nonuniqueness. This procedure uses the velocity potential of an interior point as an indicator of solution error. In all cases considered, the interior potential correctly indicated a good or bad solution, whereas the matrix condition number falsely indicated a bad solution in several instances. [Work supported in part by ONR.]

11:00

QQ9. Surface gravity wave effects on sound beams generated by a transient corrugated laser-deposited heat configuration moving at near sonic velocities. Allan D. Pierce and Hsiao-an Hsieh (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Previous articles suggest the use of a multiply split laser beam moving over the water surface at sonic velocities to create an underwater sound pulse beamed at near horizontal directions. The moving corrugated wave-train of laser-induced heating would systematically pump up the acoustic pressure and result in an improved optical-to-acoustic transduction efficiency, as well as yielding a controlled sound-radiation pattern. However, the extended nature of the effective source allows the possibility that gravity waves on the water surface may upset the phase coherence. Considered adverse mechanisms include the following: (1) the refraction of light at the tilted and curved rocking air–water interface causes the slabs of light penetrating to depths below the surface not to be perfectly parallel; (2) the rippled water surface causes the energy being added to a given moving (nearly planar) region of peak acoustic pressure to fluctuate; and (3) a nearly plane-acoustic wave propagating horizontally just below the surface is scattered by the surface ripples. Although the quantitative assessment of such mechanisms is still incomplete (and could be aided by experimental work), a rudimentary mathematical theory is given that suggests criteria for when gravity wave effects have minor influence. [Work supported by ONR.]

11:15

QQ10. Parametric study of a laser-generated acoustic signal. Luis J. Gonzalez and Ilene J. Busch-Vishniac (Applied Research Laboratories, University of Texas, P. O. Box 8029, Austin, TX 78713-8029)

The generation of underwater sound by a moving high-power laser source is numerically investigated. In particular, properties of the received acoustic signal, such as maximum and minimum periods and time-to-peak amplitude, are examined as a function of the source characteristics, such as velocity and source–receiver range and angle. Signal properties that are most and least stable to source geometry variations are quantitatively determined. [Research supported by ONR.]

11:30

QQ11. Acoustic radiation from a plate with sinusoidally varying properties. Mauro Pierucci (Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

The acoustic radiation produced by an infinite flat plate with sinusoidally varying properties and driven by a distributed load is presented. The plate is assumed to be thin with constant thickness, but with stiffness varying sinusoidally along the axis. The analysis is an extension of the work previously reported [M. Pierucci, J. Acoust. Soc. Am. Suppl. 1 **79**, S35 (1986)]. The analysis consists of solving the coupled fluid-structure equations with the added difficulty of having nonconstant coefficients. The system reduces to a convoluted equation, which has been solved analytically. The results indicate the presence of acoustic pressure components radiating in directions that are related to the difference between the forcing function wavenumber and the wavenumber of the stiffness variation. Acoustic radiation patterns will be presented.