

XXVII. COMMUNICATIONS BIOPHYSICS

A. Signal Transmission in the Auditory System

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1. BASIC AND CLINICAL STUDIES OF THE AUDITORY SYSTEM

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In cooperation with the Eaton-Peabody Laboratory at the Massachusetts Eye and Ear Infirmary we have continued our work toward understanding the auditory system and applying this basic knowledge to clinical problems.

Experimental studies of the mechanical system of the inner ear, which have been carried out in several laboratories in recent years, have mainly been concerned with motion of the basilar membrane. To determine directly the forces that produce this motion, measurements were made of sound pressure in the fluid-filled inner ear of cat.¹ These results demonstrate that (1) the middle ear provides pressure gain, (2) the system is predominantly linear, and (3) the round-window membrane acts as a highly flexible pressure release for the cochlea.

Our work on cochlear transduction has been focused on studies of the alligator lizard, where measurements of basilar membrane motion have provided an indication of the input to the mechanoelectric transduction process of the

receptor cells.² Methods have been developed which allow stable recording of intracellular potentials for a long enough interval to record responses to tones over a wide range of frequency and level.³ The results indicate that the frequency selectivity that has been observed in the responses of neurons is similar to that seen in a component of the hair-cell response and that frequency-dependent nonlinearity occurs in the hair-cell transduction.

Results from cat have led to a description of auditory-nerve responses to tones in terms of average spike rate and synchrony of the spike pattern.⁴ These results have implications both for the physiological mechanisms that are involved and for the signal processing that the central nervous system might use in discriminating sound signals. Knowledge of response characteristics of auditory-nerve fibers has been extended to some speechlike sounds.^{5,6} The application of knowledge of auditory-nerve response patterns obtained from experiments on animals has been applied to the design of "cochlear implants" for patients with profound deafness. Preliminary results show a clear improvement in the usefulness of the prosthetic devices in discriminating speechlike stimuli.⁷

Studies of the organization of the neural pathways which carry signals from the brainstem to the ear have been extended.⁸ An anatomical study, which uses a radioactive label to trace the pathways of interest, indicates that for one component of this system (the ipsilateral projection from the lateral superior olivary region) there is a clear relationship between the location in the brainstem where the nerve fibers begin and the location in the cochlea where they end. This result is consistent with the view that this portion of the system has roughly the same frequency-to-space mapping in the afferent and efferent neural endings in the cochlea. The other components of the olivo-cochlear system are organized differently.

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B. Auditory Psychophysics and Aids for the Deaf

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1. INTENSITY PERCEPTION AND LOUDNESS

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This research is concerned with the development of a unified, quantitative theory of intensity perception and loudness, and involves the construction and integration of models of sensory processes, short-term memory, perceptual-context effects, and decision making, as well as extensive psychophysical experimentation. During this period our work can be divided into four main areas: (i) work on the new context-coding model, (ii) loudness discrimination between stimuli with

different dynamic ranges, (iii) intensity discrimination in listeners with hearing impairments, and (iv) preliminary work on multichannel discrimination.

(i) The new model of context coding is addressed to the observed increase in resolution that occurs near the edges of the range when unidimensional stimuli are identified (or discriminated in the context-coding mode) and that is believed to result from the use of perceptual anchors near the sensations corresponding to the extreme stimuli in the range. Although we have developed a precisely formulated model for this effect based on the notion of measuring the sensation relative to noisy perceptual anchors with a stochastic ruler, we have not been able to determine analytical expressions for the conditional densities of the decision variable, and we have been forced to derive the predictions of the model computationally. During the past year we have focused on the problem of evaluating the parameters of the model and of deriving insight into the form of the decision densities in order to understand the predictions of the shapes of the associated ROC's.

It appears that a wide variety of large-range identification data can be fit by the model with very uniform parameters. Specifically, the choice of an average measurement step size equal to 0.03 of the sensation range, anchor jitter precisely equal to the sensation noise, and mean anchor positions offset by one standard deviation away from the extremes of the sensation range fits a wide variety of results on large-range tone-identification experiments¹⁻⁴ and a large-range noise-identification experiment⁵ as well as being consistent with the results obtained in small-range identification experiments.⁶ The results reported for an experiment in which stimulus range was varied systematically are somewhat better fit if it is assumed that anchor jitter is twice as large as the standard deviation of the sensation noise but they appear to be at variance with the small-range results reported for the small-range experiment.

For this range of parameters, the ROC's predicted by the model are very nearly straight lines whose slopes differ from unity by no more than 10%. This is surprising because it can be shown that the variance of the decision densities in the middle of the range is roughly a factor of 10 larger than that of those corresponding to the extreme stimuli. In order to understand this apparent discrepancy, we have examined the ROC's corresponding to Edgeworth approximations to the

decision densities of various orders. When the effects of the skew and kurtosis of densities are accounted for, the predicted slope discrepancy associated with the change in variance practically vanishes within the (0.01-0.99) confines of the observable probability range.

A manuscript on the new context-coding model and a brief note on this application of the Edgeworth approximation technique are being prepared for publication.

(ii) Preliminary work on the variability of loudness comparisons utilized a roving-level, two-interval, two-alternative symmetric forced-choice discrimination paradigm with tonal stimuli differing in frequency.⁷ To avoid use of the context-coding mode for loudness comparisons, frequency differences were kept small (1000-600 Hz) and, consequently, the dynamic ranges of the stimulus types were essentially the same. In order to study the variability of loudness comparisons in listeners whose hearing impairment includes a reduction in dynamic range such as that normally associated with recruitment, we have begun to study the variability of loudness comparisons between pure tones and partially masked tones of the same frequency in listeners with normal hearing. To avoid problems of temporal masking and the fatigue caused by continuous exposure to the masking noise, the comparison involves presenting the pure tone to one ear and the masked tone to the opposite ear. Five roving-level, two-alternative symmetric forced-choice experiments have been conducted. Two are monaural experiments involving comparisons between the same stimulus types, two involve interaural experiments between the same stimulus types, and one involves interaural comparison between different stimulus types. This series is needed to separate out effects associated with interaural comparisons from those associated with comparisons between different stimulus types. Both kinds of effects are likely to be encountered when a unilaterally impaired listener makes loudness comparisons between a normal ear and an impaired ear.

A preliminary analysis of the results indicates that the variance introduced by the interaural comparison is roughly independent of stimulus type. The size of this variance is slightly larger than the sum of the sensation and memory trace variances found in monaural comparisons.

(iii) Current studies of intensity discrimination in listeners with hearing impairments include further tests of impaired listeners, tests of normal-hearing

subjects with simulated losses, and the development of improved analysis procedures. During the past year we have focused on the measurement of intensity discrimination in normal-hearing listeners with simulated hearing losses.

When pooling intensity-discrimination results obtained from different impaired listeners and when comparing these results with those obtained from listeners with normal hearing, it is important to consider how best to normalize the results with respect to differences in test frequency and to variations in the relation of the test frequency to the characteristics of the audiogram. In addressing this problem, we are not only including listeners with normal hearing for the usual control purposes, but are also testing the same normal-hearing listeners under conditions that simulate the losses of the impaired listeners. This procedure not only provides us with an additional means of characterizing the perceptual performance of the impaired listeners (i.e., their performance can be described in relation to that of the simulations as well as in relation to normal performance), but also provides additional subjective insight into the perceptual effects of impairments (by serving as experimental subjects ourselves), and, more generally, provides an increased background for the development of theoretical models.

In general, our ability to simulate sensorineural impairments by altering the input signal (i.e., by attenuating, filtering, distorting, adding noise, etc.) applied to a normal listener is quite limited. Not only are there fundamental difficulties in achieving simulations that are subjectively realistic or objectively accurate over a wide variety of tasks, but relatively little is currently known about useful techniques for achieving adequate simulations even for highly restricted situations. Despite these limitations, however, we believe that simulations can be highly useful. At worst, they provide an interpolation between the impaired case and the normal case, and permit certain hypotheses about the performance of impaired listeners to be rejected.

Currently, most of our simulation work is based on the use of shaped masking noise. Although this form of simulation obviously has strong limitations, it is superior to the absence of any simulation, and to the simulations that are implicit in performance comparisons at equal SPL's, equal SL's, or equal loudness, or even to simulations achieved through the use of attenuation that is level-dependent as well as frequency-dependent. For example, in cases in which the

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impaired listener has a normal discomfort threshold curve, it can (unlike frequency-dependent and level-dependent attenuation) produce reasonably accurate results not only for the boundaries of the auditory area and the growth of loudness within this area, but also for the discrimination of intensity within this area. It appears that the performance of our impaired listeners can often be reasonably well approximated by this form of simulation and that, in general, the results for impaired listeners can be more usefully organized when the simulations are used as a comparative reference than when normals are used for this purpose. In other words, the intersubject variance is significantly decreased by first normalizing with respect to the simulation.

A further project now under way is concerned with determining the class of audiograms that can be well approximated by the use of masking noise and with the extent to which different masking spectra can lead to approximately the same audiogram. Clearly, both of these questions are important in considering the usefulness of this form of simulation. At present, attention in this project is focused on the problem of simulating high-frequency losses with steep slopes. The most direct approach, using a broadband high-frequency noise with a sharp cutoff but a continuous spectrum, typically results in significant masking below the cutoff frequency and limits the severity of the loss that can be simulated. An alternate approach, which reduces the masking below the cutoff but leads to a somewhat non-uniform threshold at high frequencies (rippled high-frequency audiogram), employs a multiplicity of narrow-band high-frequency noise maskers. We are presently studying the limitations of both of these approaches in some detail.

Current work also includes an attempt to organize the results of our work and past work reported in the literature into a coherent overview of intensity resolution in impaired listeners. This work includes a review of results on the SISI Test,⁸ and an attempt to relate the results obtained with this procedure to procedures such as ours, e.g., that require comparison of pulsed tones.⁹⁻¹²

(iv) Work on multichannel discrimination is concerned with experiments in which a listener must discriminate between stimuli which may differ in more than one perceptual dimension, such as experiments involving discrimination between spectra of different shapes. If the experiment is conducted in the standard manner, using a single representation of each stimulus type, the discrimination can

be accomplished using information coded in only a single dimension (e.g., a single critical band) or by using largely irrelevant cues (e.g., loudness) rather than by comparing the information available in the plurality of dimensions properly (e.g., to abstract shape). Although it is, in principle, possible to eliminate extraneous cues by properly "balancing" the stimuli, this approach has obvious practical and theoretical limitations.

An alternate approach is to rove the value of all stimulus parameters irrelevant to the intended discrimination pattern. For example, for spectral shapes with differences in two frequency bands roving overall level eliminates both overall loudness and loudness within a single band as reliable cues, and forces the discrimination to be made on the basis of the relative loudness of the two bands. Shapes involving differences in three bands would require random variation in spectral tilt, as well as overall levels, to assure that discriminations were not being made on the basis of loudness in two bands alone.

We have begun to structure a model for the performance of observers in such multidimensional-discrimination experiments. According to this model, each dimension corresponds to an independent perceptual channel having limited sensory resolution. The observer attempts to discriminate between stimuli by computing a decision variable that normalizes out, in an optimal fashion, the effects of the irrelevant stimulus variations. We have begun to determine the effects of range of variation and number of channels on the performance of such observers. It is apparent that, for a given number of channels, there are optimum stimulus patterns for discrimination prototypes. Further, an observer with a fixed number of channels should be unable to discriminate between stimulus patterns when the number of irrelevant variations exceeds the number of available channels. Experiments on spectral-shape discrimination are currently being planned to help guide the development of this model.

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2. BINAURAL HEARING

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Our overall goal is to understand binaural phenomena in normal and impaired listeners and to relate these phenomena to physiological data whenever possible. This goal is approached by a combination of empirical and theoretical work. For the purposes of this report, our activities are divided into studies of binaural interaction in normal listeners and studies of binaural interaction in impaired listeners.

Binaural Interaction in Normal Listeners

In an extension of our previous empirical work on correlation discrimination,¹ we are completing a theoretical treatment of the relation between correlation discrimination and masked detection. Preliminary calculations show that observed NOSII detection thresholds are within 1-2 dB of the values predicted from interaural correlation jnds (measured from a reference condition of +1). In contrast, we find predicted thresholds which are 9 dB higher than observed thresholds when we compare NUSO detection to thresholds predicted from correlation jnds from a reference correlation of 0. Apparently, interaural correlation changes may mediate detection for the NOSII signal configuration but, clearly, do not mediate detection for the NUSO configuration. A paper describing this work is in draft form.²

We completed the study of auditory localization reported in last year's progress report (RLE PR No. 122). In this study, performance with a localization simulator (using headphones) is compared to performance with real sources in an anechoic chamber. The final results and analysis³ support the preliminary indications that performance with simulated sources is worse than performance with live sources, at least for vertical-angle (elevation) judgments. A possible source of the discrepancy is that the insert microphones do not adequately record the signals that would be present in the ear canals in the absence of the microphones, either because they are too large or because they are not far enough inside the ear canal to avoid significant influence of transverse vibrational modes. A conclusive test of this hypothesis should require relatively little additional study.

Our study of interaural time discrimination of a narrow-band noise in the presence of a wide-band noise⁴ was also extended with a theoretical modeling effort. Predicted jnds for ten different configurations of the masking noise were computed and compared with empirical results using a model based on the instantaneous interaural time delay and interaural intensity difference of the total signal. In contrast to other attempted explanations (e.g., those based on relative detectability or those based on relative lateral position), this formulation gives rough agreement between theory and empirical results.

We have also pursued a theoretical study of a neural mechanism for the

processing of interaural intensity differences. Early work⁵ shows that the elements of the model behave in a manner consistent with data from neurons in the lateral superior olivary region and with psychophysical observations.

Binaural Interaction in Impaired Listeners

Our experimental program studying binaural interaction in hearing-impaired persons includes a survey with a number of crude tests on many subjects, as well as in-depth studies with careful tests on a small number of subjects.

In the past year, much of our effort has been invested in preparing our anechoic chamber facility for the free-field experiments that are part of the survey of binaural interaction in hearing-impaired listeners. We have purchased and installed a new computer system, a PDP 11-23. We have constructed appropriate interface electronics to allow computer control of the programmable analog equipment (e.g., attenuators, electronic switches). We have developed programs to provide stimulus and response control for experiments in spatial discrimination, absolute identification of directions, and detection. These programs provide a graphic display of the accumulated data for the current test, assist the experimenter in choosing the next stimulus, and record and process responses. The stimulus is presented from a loudspeaker mounted on a boom, the position of which is controlled by the experimenter. The subject-chair-boom system has been mounted onto a stable platform that rests on the cement subfloor of the chamber. With this arrangement, the stimulus is decoupled from any motions of the wire-grid floor on which the experimenter is standing. We have tested our procedures on normal and impaired listeners and are ready to run larger numbers of subjects.

For our in-depth studies⁶ the subjects are being chosen from three classes of impaired listeners with bilaterally symmetric hearing: listeners with noise-induced high-frequency losses, listeners with flat losses, and listeners with multiple sclerosis who have normal pure-tone audiograms. Each subject is tested on four fundamental measures of binaural interaction: interaural time discrimination, interaural amplitude discrimination, interaural correlation discrimination, and binaural masked detection. Discrimination jnds and detection thresholds are measured for several reference conditions and several signal-masker configurations. Stimuli in the discrimination experiments are third-octave band

noises at six center frequencies. The same stimuli are used as maskers for tonal targets in the detection experiments. Early results have been interesting. For example, the first subject studied with multiple sclerosis is unable to discriminate between correlations of 1 and 0 (whereas normals discriminate between 1 and 0.96). We believe this study will also help us to understand binaural interaction in normal listeners. For example, consider a subject with a noise-induced high-frequency loss who shows reduced release from masking (approximately 6 dB compared to approximately 15 dB for normals) in both the normal and impaired regions of his audiogram. This subject's interaural correlation jnds, which are also poorer than normal, show the same relation to his detection thresholds as observed for normal listeners (discussed above).

In an intensive study of a single hearing-impaired subject, we performed many psychophysical tests on a subject with a well-documented family history of sensorineural hearing loss.⁷ Our tests included rough measurements of absolute threshold, equal-loudness contours, amplitude jnds, frequency jnds, psychophysical tuning curves, binaural detection thresholds, interaural correlation jnds, interaural time jnds, interaural intensity jnds, vertical-angle discrimination, pitch matching, and CV syllable identification. Although the psychophysical data did not support any simple physiological interpretation of Rapoport and Ruben's findings,⁷ comparisons of the results of different measurements suggested theoretically meaningful relationships between psychophysical measurements. Unfortunately, our subject relocated away from the Boston area so that these relationships could not be pursued with additional tests or more precise measurements. The measurements that we did make, however, support the notion⁸ that an extensive, carefully planned testing program on one impaired listener can generate significant insight into auditory function and modelling.

Finally, we have developed a sequential estimation procedure which can significantly reduce the amount of time required to collect experimental data.⁹ This should be an important tool for our experimental work with hearing-impaired listeners. The procedure is nonadaptive, and hence does not alter stimulus values during the course of a run. Instead, the procedure inspects a subject's cumulative percent correct at the end of each trial and, based on a set of well-defined criteria, decides whether the subject is near the level of performance specified

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by the experimenter. If so, the procedure continues presenting trials and recording responses. If not, the procedure stops the run and records the percent correct. If, at the end of a specified number of trials (in our case, 50), the procedure has not stopped, the run is terminated and a percent correct estimated. In our experience, this procedure provides approximately a 50% savings in time over the fixed-number-of-trials procedure.

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3. HEARING AID RESEARCH

National Institutes of Health (Grant 5 R01 NS12846)

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This research is concerned with the development of improved speech-reception aids for persons suffering from hearing loss, and the improvement of fundamental understanding of the limitations on such aids. Our work in this area is focused on the problem of developing improved signal-processing schemes that match speech to residual auditory function for listeners with sensorineural impairments. During the past year we have continued to study schemes based on (i) linear amplification, (ii) multiband amplitude compression and (iii) frequency lowering, and have continued to study (iv) the effects of speaking clearly to improve communication with the hearing impaired.

(i) Research on linear amplification is primarily concerned with evaluating the extent to which Articulation Theory^{1,2} can be used to predict the dependence of speech intelligibility on frequency-gain characteristic and presentation level for listeners with sensorineural impairments. A study is under way to determine the relationship between actual and predicted performance-intensity (PI) functions for a variety of filtering conditions and presentation levels. Six subjects with normal and impaired hearing have been tested. Three subjects have bilateral sensorineural hearing losses of varying degrees and etiology, one subject has a unilateral sensorineural loss, and two subjects have normal hearing. The nine filtering conditions include high-pass filtering at 700 Hz, 1400 Hz, and 2800 Hz, low-pass filtering at 700 Hz, 1400 Hz, and 2800 Hz, and bandpass filtering at 700-1400 Hz, 1400-2800 Hz, and 700-2800 Hz. All filtering is accomplished with linear-phase filters having steep rejection characteristics. Each of the filtering conditions and the unfiltered speech were presented to each subject at a minimum of five different hearing levels. For the normal-hearing subjects, presentations were made

both with and without simulated hearing loss. Hearing loss in the normal-hearing subjects is simulated by using spectrally shaped wide-band noise to create masked thresholds similar to those of the hearing-impaired subjects.

The obtained PI functions for the conditions reflect the amount of speech-spectrum information available. For example, for speech low-pass-filtered at 1400 Hz, maximum performance from normal-hearing listeners with no simulated hearing loss is about 60% at a presentation level of 60 dB HL. Introducing noise to simulate a hearing loss shifts the PI function upward in intensity. Higher presentation levels are necessary to overcome the masking of the noise. Although the slope of the PI function is generally greater in the simulation case than in quiet, the maximum performance achieved is roughly the same. For most of the impaired subjects tested, the slope of the PI curve appears to be similar to that for the simulated-loss case. However, while performance at lower presentation levels is comparable to the simulated-loss case, maximum performance is generally poorer, and there is a greater tendency for performance to decrease at higher levels.

A significant observation for one subject with bilateral noise-induced high-frequency hearing loss is a marked improvement in performance for the high-pass-filtered speech at 700 Hz compared to performance for unfiltered speech. Except for requiring higher presentation levels, his performance is comparable to that achieved by normal-hearing listeners for this condition. Other hearing-impaired listeners have not benefitted from this filtering condition. For example, one listener with the unilateral hearing loss achieved much poorer scores in her impaired ear than in her normal ear with simulated loss under this condition.

Efforts are now under way to compare these experimental results with the predictions of articulation theory.

(ii) Research on amplitude compression for listeners with reduced dynamic range has included efforts to analyze the performance of multiband compression systems as well as an initial study of multiband limiting.

Amplitude distributions which probabilistically describe the level variations present in third-octave bands during speech have been obtained through the periodic sampling of the short-term rms speech level in each band. These density functions have been derived for a comprehensive set of speech materials, including

isolated CVC syllables and nonsense sentences spoken by male and female talkers.³ In general, the amplitude distributions do not have simple analytic forms, although they can be described parametrically in terms of cumulative percent levels. Typically, the range of amplitudes between the 10% and 90% cumulative levels exceeds 40 to 50 dB. Detailed characteristics of the density functions, however, change significantly across frequency. In the low-frequency bands, the distributions are bimodal, reflecting distinct amplitude ranges for voiced and unvoiced speech segments. The distributions become unimodal and more peaked in the higher frequency channels. These results are inconsistent with previous characterizations of speech-level distributions as being roughly 30 dB wide and uniformly distributed in all frequency ranges.^{1,4} Evaluations of these discrepancies are currently under way.

A study of amplitude distributions for unprocessed and compressed materials has been conducted in order to analyze the behavior of multichannel compression systems.⁵ In general, since compression is a nonlinear process with memory, these systems are defined in terms of their response to simple static and dynamic stimuli. However, these descriptions are inadequate to predict the processing of a more complex signal, such as speech. Analysis of the processed level distributions has led to a simple model of multichannel compression which incorporates the interactions of static and dynamic system properties with the dynamic characteristics of speech. The study indicates that the static compression ratio applies only to components of the speech envelope that vary on a syllabic time scale. Dynamic variations which occur at a rate comparable to or faster than average pitch periods are essentially uncompressed. As a consequence, the range of output levels typically exceeds that predicted on the basis of the static compression characteristic.

A similar analysis of compressor behavior has been completed⁶ for the system hardware used in previous studies of compression.⁷⁻⁹ The results of this analysis indicate that the initial negative findings for multichannel compression are not significantly attributable to limitations in the system implementations. In addition, this study provided measures of the speech-level distributions produced by several previously investigated linear and compression systems under conditions which replicated the original experiments with two impaired subjects. These

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distributions allow the comparison of intelligibility test results with the actual range of amplitude levels present between the subject's detection and discomfort thresholds. Preliminary analyses suggest that surprisingly high levels of intelligibility were achieved when only small amounts of speech were audible.

An evaluation of multiband limiting, designed to protect hearing-impaired listeners from high-level speech and environmental sounds, has been conducted. The relative performance of two linear and four compression-limiting systems has been measured for two subjects with flat sensorineural losses. The test materials, consisting of nonsense CVC syllables spoken by male and female speakers, were presented over a 30 dB range of input levels. The subjects chose an overall listening level consistent with both maximum intelligibility over the range of input levels and long-term listening comfort. Preliminary results indicate that both average presentation levels and increased speech-intelligibility scores are associated with compression limiting relative to linear amplification.

Averaging over all the presentation levels, the compression-limiting intelligibility scores demonstrated a small (4%) advantage over those of the linear systems. The compression-limiting systems studied may be grouped into two categories: (1) the less severe compression (LSC) systems, those with compression thresholds at 10 or 20 dB below the subjects' discomfort levels and (2) the more severe compression (MSC) systems, those with compression thresholds 10 dB below those of the LSC systems. The performance of the LSC systems was as good as or better than that of the corresponding linear systems at all of the input levels. Intelligibility scores for the MSC systems were much better than for either the LSC or linear systems at low levels but dropped sharply below the others at the higher levels. These trends were independent of the frequency-gain characteristics.

(iii) Research on frequency lowering for listeners with negligible hearing at high frequencies has continued to focus on pitch-synchronous time-dilation techniques that can incorporate warping of short-term spectra. During the past year we have developed two frequency-lowering techniques based on linear prediction of the speech waveform.¹⁰ In one system, linear prediction is applied to frequency-lowered discrete power spectra.¹¹ In the second, the poles of the LPC synthesis filter are relocated within the unit circle. The intelligibility of

materials processed by these methods is currently being compared with that of materials warped and lowered by the scheme extensively studied in previous work.¹²

(iv) Research on the effects of speaking clearly on the intelligibility of speech for impaired listeners has been expanded to include both sentence-level and segmental-level phenomena. In one study,¹³ measurements of the intelligibility of clearly and conversationally spoken nonsense sentences were made for 8 normal-hearing listeners listening to unprocessed speech and speech sharply low-pass-filtered at 1 KHz. For unprocessed speech, the intelligibility of both clear and conversational speech was high (above 95% correct) while for low-pass-filtered speech, clear speech was substantially more intelligible (average of 13 percentage points). In another study,^{13,14} the intelligibility of plosive consonants in CV contexts spoken clearly and conversationally was measured for normal-hearing listeners listening at various S/N ratios. For two of the three speakers, plosives spoken clearly were much more intelligible than plosives spoken conversationally at a variety of S/N ratios (13 percentage points). Acoustic analysis of the plosives indicated that differences in VOT between voiced and unvoiced plosives were much larger in clear speech, consistent with better perception of the feature voicing. Large differences in plosive burst frequencies and CV ratios were not seen, although a detailed examination of intelligibility results revealed some correlation between higher intelligibility and burst frequencies moving closer to target values in clear speech. Finally, a major study on the intelligibility of conversationally and clearly spoken nonsense sentences for hearing-impaired individuals is nearing completion. Preliminary results indicate that differences in the intelligibility of clear and conversational speech (18 percentage points) are relatively insensitive to manipulation of overall level and frequency-gain characteristic. Work to be pursued over the coming year includes further analysis of segmental-level data obtained on plosives, studies on the effects of overall rate on clear and conversation speech, and the perception of clear and conversational speech by normals listening to speech degraded by noise and filtering.

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4. TACTILE COMMUNICATION OF SPEECH

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The goal of this research is to develop tactile speech communication aids for the deaf and deaf-blind. Our past research on the tactile communication of speech can be divided into six main areas: Study of Tadoma, Development of a Synthetic Tadoma System, Experiments on Optacon-Based Spectral Displays, Development of a New Stimulator Array, and Display of Vocal-Tract Area Functions.

In Study of Tadoma, we have surveyed Tadoma use in the United States (using both questionnaires and in-depth interviews), performed a wide variety of experiments on a number of experienced, deaf-blind Tadoma users (concerned primarily with speech reception, but also including speech production and linguistic competence), and conducted a number of experiments on normal-hearing and sighted subjects having simulated deafness and blindness (concerned primarily with the perception of isolated speech segments, and secondarily with the perception of sentences). The results of this work have demonstrated that speech communication (and the learning of language) is possible through the tactile sense, and that the ability of experienced, deaf-blind Tadoma users to understand speech through the tactile sense is not based on exceptional tactile sensitivity. This work has also begun to help us characterize the perceptual cues used in Tadoma, the amount and types of training required to learn Tadoma, and the similarities and differences between Tadoma and other (artificial) methods of tactile communication.

In Development of a Synthetic Tadoma System, we have (with the aid of the Speech Communication Group at MIT) begun to obtain multichannel recordings of signals generated by the facial actions associated with speech production that we believe are crucial in Tadoma (e.g., laryngeal vibration jaw movement, lip movement, air flow), and also to develop the artificial face that will be driven

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by these facial-action signals and serve as the display in our synthetic system. Current work on this artificial face (which makes use of a model of the human skull) is focused on the design of the lip-movement system and the completion of the jaw-movement system. The adequacy of this system will be tested by comparing speech-reception results obtained with this system to results obtained with "real Tadoma." Once an adequate synthetic system has been developed (a goal that we hope to achieve within the next two years), it will be used as a research tool to explore perceptual effects of transformations in Tadoma that cannot be achieved with real Tadoma. These transformations will include both (1) alterations of the input information (e.g., adding physical cues normally not available or eliminating cues that normally are available) and (2) alterations of the display while holding the input information constant (e.g., simulating the facial output on a vibratory array). In some cases, these transformations will permit more refined and carefully controlled tests of alterations already considered with real Tadoma (e.g., changes in rate, selective elimination of voicing, jaw movement, lip-movement, or airflow, etc.).

In Experiments on Optacon-Based Spectral Displays, we performed a number of experiments on the discrimination and identification of speech segments using a computer-controlled Optagon array applied to the index finger. The results of these experiments suggest, at least for the special cases tested, that static and time-swept displays do not lead to radically different performance and that Optacon-based spectral displays are generally not very effective (e.g., compared to Tadoma or to the MESA system developed by Sparks et al.¹).

In Development of a New Stimulator Array, we have constructed a new rectangular vibratory array which includes 81 to 256 piezoelectric bimorph benders (depending upon the size of the bimorphs employed), a mounting and suspension system to conform the array to the skin surface, activation and control electronics to excite the array, and software to facilitate specification of vibrating patterns on the array. This array, which is derived from the array developed at Princeton (Sherrick and Geldard²), is considerably more flexible than the Optacon (it can be applied to a wide variety of body surfaces, it allows excitation frequency and wave shape to be varied, and it permits the relative amplitude and phase of each element to be controlled individually). By employing it, as well

as the Optacon, in our research, we hope to gain further insight into the limitations encountered with Optacon-based displays.

In Display of Vocal-Tract-Area Functions, we have initiated research on the relative usefulness of displaying vocal-tract-area functions as opposed to short-term spectra (a research direction also currently being pursued by M.H. Goldstein at Johns Hopkins University and by J.C. Craig at Indiana University). Although vocal-tract-area representations of speech and certain spectral representations of speech are mathematically equivalent, it is possible that the former may be more suitable for tactile communication. In particular, the vocal-tract representation is more directly related to the production process (so that the associated displays may be easier to learn) and involves less abrupt changes across phoneme boundaries (e.g., when area functions are linearly interpolated across such boundaries, the resulting synthesized speech is superior to that which is achieved by performing the same operations with spectra (see Shadle and Atal³)). Research to date in this area has focused on mathematical analysis and the development of techniques for transforming acoustic waveforms to tactile patterns. Experiments are now being initiated comparing a vocal-tract-area display, a spectral display, and Tadoma. In the vocal-tract-area display, the contours are represented directly by spatial patterns, the level is represented by the intensity of stimulation, and the vocal-tract excitation information by the characteristics of the stimulating waveform.

In Basic Study of Encoding and Display Systems, we have initiated an attempt to structure the problem of encoding and displaying time-varying contours (e.g., a spectrum or a vocal-tract-area function) in a manner that will lend itself to more systematic exploration. In general, the time-varying contour is represented by a set of time functions, and the encoding and display scheme is defined as a real-time mapping of the values of these functions into tactile patterns on the array. In initial work, we are assuming, for simplicity, that the functions are two-valued; that the task is to detect, discriminate, and identify a segment of only one such function; and the region of the array devoted to displaying this single function is restricted in a manner that permits expansion to the whole set of functions later on. The portions of the specified function outside the given segment to be perceived provide realistic temporal surround, and the portions of the array outside the specified region provide realistic spatial surround. Current

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work is concerned with the selection of a subset of encoding and display schemes for experimental exploration (even with the above simplification, the number of possible schemes is enormous) and with the detailed design of experimental procedures.

In Preparation of Results for Publication, we have prepared five new journal articles: Reed et al.,⁴ concerned with Tadoma performance by an experienced, deaf-blind user; Reed et al.,⁵ concerned with Tadoma performance by normal subjects; Snyder et al.,⁶ concerned with a comparative study of Tadoma and a spectral display; Clements et al.,⁷ concerned with a comparative study of two spectral displays; and Reed et al.,⁸ concerned with a general review of recent results on tactile communication of speech. Further articles now being prepared include the results of the Tadoma survey, more extensive results on speech reception by Tadoma users, and the results on the linguistic competence of these users (some of whom have been deaf and blind since infancy).

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5. MUSICAL PITCH

National Institutes of Health (Grant 2 R01 NS11680)

Klaus J. Buhkert, Adrian J.M. Houtsma

The overall objective of this research is to understand the process of musical pitch sensations evoked by complex sounds. Efforts have been focused on the following projects.

a. Pitch of Harmonic Two-Tone Complexes

The investigation of the effect of component intensity on the pitch of two-tone complexes was completed.¹ Dichotically presented two-tone complexes with inter-tone intensity ratios of 0, ± 10 , and ± 20 dB and frequencies nf_0 and $(n+1)f_0$ were used to play melodic intervals to be identified by several subjects. Empirical confusion matrices were correlated with theoretical matrices computed from Goldstein's optimal-processor theory, Wightman's pattern-transformation theory, and from an analytic pitch theory which assumes that a two-tone complex has one of two possible pitches, corresponding to the spectral frequencies. Results indicate that the best predictions are achieved by a combination of the optimum processor and the analytic theories. At low harmonic numbers, the optimum processor alone accounts very well for the data, while at higher harmonic numbers a combination of both theories yields the best correlation with observed data. At harmonic numbers higher than about six, correlation between data and any theory or combination of theories is less than unity, suggesting that pitch behavior for complex tones of high harmonics has some randomness currently not accounted for by any theory.

b. Pitch of Amplitude-Modulated Noise

Periodically interrupted white noise has a flat long-term power spectrum and weak pitch properties, whereas periodically repeated white noise has a definite

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line spectrum and evokes a strong pitch sensation. Musical-interval recognition experiments were performed using periodically gated digital noise, where the gate rate corresponded to the notes to be played, and the correlation between corresponding samples in successive gates was systematically varied between zero and unity. Zero correlation between samples leads to a signal equivalent to interrupted white noise, whereas unit correlation results in a periodically repeating waveform. Signals were band-passed, and performance was measured as a function of bandpass center frequency and sample correlation. Data analysis and modelling attempts are currently in progress.

c. Induced Pitch Shifts for Pure and Complex Tones

In an earlier study, we reported that noise-induced shifts in the pitch of pure tones are considerably larger than the shifts induced in the apparent complex-tone pitch or "residue."² This finding seemed inconsistent with the notion of serial pitch processing (tone frequencies \rightarrow spectral pitches \rightarrow "virtual" or "Gestalt" pitch), and it was suggested that pure tone pitch and complex tone pitch are processed through separate channels, the former based on spatial, the latter on temporal encoding. Several control experiments were performed involving pure-tone pitch shifts induced by contralateral noise, an attempt to repeat results that were obtained by another investigator and which seemed to contradict our results,³ and an experiment suggested by Terhardt² in which noise-induced pure-tone and complex-tone pitch shifts were measured simultaneously using one complex stimulus. Contralaterally induced pure-tone pitch shifts were of about the same (small) magnitude as noise-induced shifts in the pitch of complex tones. This suggests that there are two independent pitch-shift mechanisms which have a cumulative effect. One is rather peripheral and affects only pure-tone pitch; the other is central and affects both pure-tone and complex-tone pitch. A report is currently being prepared for publication.⁴

d. Tactile Music

We have begun to study the feasibility of making music available to the deaf-blind through tactile stimulation. The long-term goal of this project is to build a system that will accept an acoustic music input, extracts relevant

parameters in real time, and presents stimuli derived from these parameters to an array of vibrators attached to the skin of the "listeners." Initial studies are focused on finding tactile correlates to basic musical phenomena such as consonance, dissonance, rhythmic and melodic pattern perception, and chords.

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C. Transduction Mechanisms in Hair Cell Organs

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The overall objective of this project is to study the sequence of steps by which mechanical stimuli excite receptor organs in the phylogenetically related auditory, vestibular, and lateral-line organs. The receptor cells in these organs are ciliated hair cells. Specific goals include the characterization of the motion of the structures involved, particularly the hair-cell cilia; study of the nature and origin of the electrical responses to mechanical stimuli in hair cells; and investigation of the role of these responses in synaptic and neural excitation.

Goals during the current year have been mainly in two areas: (1) to measure the mechanical response of a simple auditory organ, with particular attention to the way in which the response pattern depends upon frequency and (2) to characterize the semicircular canal mechanical response by direct measurements and through theoretical considerations.

1. MECHANICAL RESPONSE PROPERTIES OF THE BASILAR PAPILLA IN ALLIGATOR LIZARD: FAILURE TO FIND A BASIS FOR TONOTOPIC ORGANIZATION

National Institutes of Health (Grant 5 R01 NS11080)

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Observations of receptor potentials and neural responses from cells of the auditory receptor organ (basilar papilla) of the alligator lizard indicate that the organ is tonotopically organized, i.e., frequency response depends on location along the basilar membrane.¹ However, mechanical measurements, employing the Mossbauer method, suggest that there is no frequency response dependence of the basilar membrane motion on position along the membrane.² We have attempted to observe the motion of the papilla and of the hair-cell cilia optically, hoping to find

some frequency-selective response as a basis for the physiological results. Under stroboscopic illumination we have examined the motion of the papilla in response to tones in a preparation excised from the animal. The results are consistent with those obtained earlier by the Mossbauer method, i.e., the entire papilla moves in phase and there appears to be no spatial dependence of the response on frequency over the range of 500 Hz to 4000 Hz. The papilla rocks about its long axis, displacing the ciliary tufts in the direction of maximum sensitivity, as inferred from their morphology. The vertical motion of the basilar membrane on which the papilla rests is probably somewhat larger than the lateral rocking motion of the papilla but cannot be accurately measured in the microscope. This is the component of motion that was measured by the Mossbauer method. Thus neither of these two (orthogonal) modes of motion provides a basis for the observed tonotopic organization of the papilla. The mechanical responses of the cilia may hold the answer to this riddle and studies of their motion are in progress.

Attempts to study the motion of the basilar papilla in situ have been defeated by the absence of an adequate illumination path for transmitted light. In reflected light, an in situ preparation may be feasible; we are currently exploring this and other possible approaches to the problem.

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D. Biomedical Engineering

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Included under this heading are a variety of topics in biophysics, physiology, and medical engineering. Many of these are individual projects of students supported by a training grant from the National Institutes of Health.