XXVII. COMMUNICATIONS BIOPHYSICS

A. Signal Transmission in the Auditory System

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1. BASIC AND CLINICAL STUDIES OF THE AUDITORY SYSTEM

National Institutes of Health (Grants 5 P01 NS13126 and 5 K04 NS00113, and Training Grant 5 T32 NS07047)

Nelson Y.S. Kiang, William T. Peake, Thomas F. Weiss

In conjunction with the Eaton-Peabody Laboratory at the Massachusetts Eye and Ear Infirmary we have reported studies concerned with stages of the auditory system from the inner ear to the brain stem.

Inner ear studies, which extend our knowledge of mechanical-to-neural transduction, have primarily involved experiments with alligator lizard. Measurements of basilar membrane motion indicate that tonotopic organization is not present and that the frequency dependence of the basilar membrane motion is similar to that of the stapes motion.[9] Intracellular potentials recorded from the receptor organ of the inner ear in response to clicks can be classified in two categories which probably correspond to receptor cells and supporting cells.[1] Receptor cell responses have large oscillations, which reflect the frequency selectivity of the receptor potential, superimposed on slower depolarizing potentials, which reflect nonlinearities in the mechano-electric transduction process. Supporting cell responses have a component that is sensitive to the click rate and is apparently neural in origin. Responses of cochlear-nerve fibers from the apical end of the receptor organ have been measured in response to single- and two-tone stimuli. Results[5] show that a close relation exists between measures of frequency selectivity and two-tone rate suppression, suggesting that both arise from a common mechanism.

Recent clinical interest in the use of "cochlear implants" to provide electric stimulation to the inner ear for patients with profound deafness has led to consideration of the problems and desirable properties for such prostheses.[8] Knowledge of auditory-nerve fiber response properties with speechlike stimuli,[2] as well as basic information on the processing of speech by the system,[7] may provide guidance for such efforts. Studies with patients having multiple electrode implants (carried out at the University of Utah) indicate that even crude use of our knowledge of stimulus coding in the auditory nerve makes substantial improvements in the ability of the patients to discriminate certain speechlike stimuli, although this does not imply the ability to understand ordinary speech.[3]

The "feedback" nerve bundles that lead from the brain stem out to the inner ear have been studied in two ways. Physiological experiments have demonstrated that electric stimulation of a part of this system produces reductions in auditory nerve fiber responses which are somewhat dependent on the fibers' response properties and spontaneous firing rate.[4] An anatomical study, in which components of this efferent system were labelled with radioactive methionine, has led to a description of two components of the system which arise in distinct brainstem locations and project to distinct cochlear locations.[10]

Responses of the brain stem recorded from the surface of the skull have been studied by several groups for diagnostic use. A comparison of responses to monaural and binaural stimulation for both cat and human has indicated some similarities[6] and some differences between the responses from the two species.

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B. Auditory Psychophysics and Aids for the Deaf

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1. INTENSITY PERCEPTION AND LOUDNESS

National Institutes of Health (Grant 5 R01 NS11153-03 and Fellowship 1 T32 NS07099-01)

National Science Foundation (Grant BNS77-16861)

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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 the past year, work has been conducted in four areas: (i) the relation of intensity resolution to auditory-nerve firing patterns, (ii) the relation of intensity discrimination to masking patterns, (iii) the relation of intensity discrimination to loudness, and (iv) intensity resolution and loudness in listeners with hearing impairments.

i. We have pursued the investigation of the relation between intensity discrimination and the coding of intensity in auditory-nerve fibers. Predicted performance of reasonable models, e.g., an optimally weighted sum of the numbers of firings on individual fibers, depends critically on assumptions about the following aspects of auditory-nerve activity: the nature of randomness; the importance of the time structure relative to the mean rate of firing (count); the saturation of the rate-intensity function; the distribution of thresholds of fibers with a common characteristic frequency (CF); the distribution of CF's over fibers; the shapes of tuning curves, especially the tails; the dependence of the rate-intensity functions both on the CF relative to the stimulus frequency and on other characteristics of the fiber (e.g., spontaneous rate); and the nonlinear interactions that occur with several stimulus components (as in partial masking experiments). In addition, predictions are sensitive to the way information from different fibers is assumed to be combined centrally. Unfortunately, there exist different sets of assumptions that both correctly predict the psychophysical data and are roughly consistent with available physiological data. In spite of the theoretical latitude, many current models make assumptions that contradict available physiological data. A paper on this material has been prepared for publication.

ii. A new phenomenological model of intensity discrimination has been in its predictions compared to data from the literature. The model is based on excitation patterns derived from masking patterns[1] and is similar to that proposed by Maiwald.[2] It is assumed that (a) the excitation of each of 24 critical bands forms the input to 24 independent excitation-level discriminations — one for each critical band, (b) the individual excitation-level discriminations adhere to Weber's Law, and (c) the outputs of the excitation-level discriminations are combined through an optimum decision rule. On the basis of this model, predictions have been made for intensity discrimination of tones as a function of level and frequency and for several conditions of partial masking (including low-pass, band-pass, high-pass, band-stop, and wide-band). In general, the model fits the data in the literature better than the model proposed by Maiwald in which the discrimination is based on the single critical band with the greatest sensitivity.

iii. Work on the relation between intensity resolution and loudness matching has continued. Specifically, an attempt was made to test an extension of our theory[3] of intensity perception which predicts that two stimuli are matched in loudness when their intensities divide the respective dynamic ranges proportionally in terms of numbers of JNDs. Comparisons between our empirical results and data in the literature, particularly data on loudness matching, revealed that loudness matches depend systematically on details of the matching paradigm, and that the observed variations are roughly of the same magnitude as the precision required to test the theory rigorously. Details of this research have been submitted for publication.[4]

iv. Studies of impaired listeners[5] have focused on (a) the measurement of intensity resolution as a function of level and frequency and (b) the determination of loudness matching functions (either between ears at a given frequency or between different frequencies in the same ear). The purpose of this study is to provide careful, systematic measurements of intensity resolution in impaired listeners as well as to test our theory relating resolution to loudness matching.[3,4]

Intensity resolution was measured at a number of frequencies in the range 500-4000 Hz over essentially the whole dynamic range using a two-interval, forced-choice, symmetric discrimination paradigm with correct-answer feedback. For subjects with unilateral impairment, measurements were obtained at the same frequency in the normal and impaired ears. For subjects with bilateral losses, measurements were obtained at different frequencies in the same ear (including, when possible, one frequency with normal and one with elevated threshold). The data were processed to obtain estimates of sensitivity per bel δ'. Since δ' varies with level and it is currently unclear how to best equate levels for impaired and normal listeners, the results on δ' obtained for impaired ears were compared to the results for normal ears at equal sound-pressure levels (the SPL-comparison), equal sensation levels (the SL-comparison), and equal loudness (the L-comparison). Loudness matches were obtained using the method of adjustment (alternate binaural or alternate monaural, depending on the subject's loss). The results of these loudness-matching tests were used both to test the above-mentioned theory and to achieve the comparison between normal and impaired δ' at equal loudnesses.

Of the 11 impaired listeners tested thus far, five had unilateral losses (so that the comparison between impaired and normal could be achieved using the same subject). Four of these subjects had cochlear losses (two low-frequency, one high-frequency, and one flat loss) and one had a retrocochlear loss (mild loss with mild slope caused by surgically confirmed vestibular schwannoma).[6] The results for the cochlear cases generally showed normal resolution for the L-comparison; however, for the SPL- and SL-comparisons, the results varied from worse-than-normal to better-than-normal (with the SPL-comparison generally showing poorer resolution relative to normal than the SL-comparison), depending on the subject and/or shape of the audiogram. These cases also tended to show abnormally rapid growth of loudness. The results for the retrocochlear case showed degraded resolution for all frequencies tested and all methods of comparison (SPL, SL, or L). The growth of loudness for this case showed either exceptionally slow growth over the whole range tested or slow growth at lower levels and then normal growth at higher levels (depending upon the test frequency).

The results obtained by pooling all ears with cochlear impairments (13 ears from 10 subjects) and partitioning this pool according to audiometric configuration (low-frequency, flat, or high-frequency loss), and then comparing these results to the pool of all normal ears (9 ears from 9 subjects), showed the following. For subjects with low-frequency loss, in the SPL-comparison and L-comparison the value of δ' fell within the range of normal values; for the SL-comparison the values of δ' were larger than normal. For subjects with relatively flat losses, in the SPL-comparison δ' was below the normal range; in the SL-comparison and L-comparison δ' fell within the normal range throughout the middle portion of the dynamic range but was below normal values at both extremes of the range. For subjects with high-frequency loss, in the SPL-comparison and L-comparison δ' was below the normal range; in the SL-comparison δ' was generally within or slightly below the normal range. Among the three groups of subjects with cochlear impairments, values of δ' were generally highest for subjects with low-frequency loss, next highest for subjects with flat loss, and lowest for subjects with high-frequency loss (independent of the choice among SPL-comparison, SL-comparison, and L-comparison).

Results on the relation between intensity resolution and loudness matching show reasonable agreement between experiment and theory. In most of the cochlear cases, however, the observed result could be equally well fit by assuming that the matching functions are simply straight lines (on dB coordinates) which interpolate between the match at threshold and a match at equal SPL's near the discomfort levels. One important exception, however, occurs with the schwannoma patient. In this case, the model predicts the results at 3300 Hz (slow growth of loudness at low levels, normal growth at high levels) very accurately, and the results at 500 and 1000 Hz (slow growth of loudness over the whole range tested) very poorly.

Current work on this project includes further testing of the given subjects, testing of additional subjects, and developing improved analysis procedures. One problem that is important when pooling results and that is receiving special attention concerns the question of how best to normalize the results with respect to differences in test frequency and (more importantly) to variations in the relation of the test frequency to the characteristics of the audiogram. 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 (an area that, at least on the surface, is rather confused). This work includes a review of results on the SISI Test, and an attempt to relate the results obtained with this procedure to procedures, such as ours, that require comparison of pulsed tones.

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2. BINAURAL HEARING

National Institutes of Health (Grant 5 R01 NS10916)

H. Steven Colburn, Nathaniel I. Durlach, Kaigham J. Gabriel, Dennis M. Freeman, David T. Gallagher, Louis V. Giordano, Rudolf G. Hausler, Yoshiko Ito, Esther K. Jaffee, Peter J. Moss, Roy P. Russell, Mark E. Schaefer, Ronald A. Siegel, Carl L. Thompson

Our overall goal is to understand binaural phenomena in normal and impaired listeners and to relate these phenomena to physiological data whenever possible. During the past year, significant progress has been made in several areas, including experimental studies of normal hearing, experimental studies of impaired hearing, theoretical studies, work on facilities, and writing reports on past work.

Experimental studies of subjects with normal hearing were conducted in several projects.

First, in a continuation of work done last year, we used frozen noise waveforms to study binaural detection phenomena by comparing homophasic and antiphasic conditions in terms of the resolution and bias for individual noise waveforms. In the experimental runs, the waveform presented on a given trial was chosen randomly from a set of 10 waveforms. We found that the resolution varied considerably from waveform to waveform in both NOSO and NOSπ conditions. Therefore, the estimates of the ratio of internal to external noise powers calculated with other assumptions from results in two-interval experiments must be revised.

Second, we tested the ability of subjects to discriminate interaural correlation for different bandwidths of the noise signal (Gabriel, 1979; Gabriel and Colburn, 1980). This study is very important to theoretical development since most theories of binaural hearing are (or can be) stated in terms of interaural correlation operations; these data allow direct tests of the basic postulates of the models. One surprising results is that discrimination performance from a reference correlation of unity deteriorates as bandwidth is increased beyond the critical band, even when the noise power per cycle is constant (fixed N_0). In other words, the subjects are apparently unable to restrict attention to a critical band in this experiment; if they could, performance would not deteriorate because the characteristics of the internally processed signal would not change for supercritical bandwidths. In another correlation study, we measured the abilities of a common set of subjects to discriminate interaural correlation and to detect target bands of noise in analogous cases (Gallagher, 1979; Gabriel, Gallagher, and Colburn, 1980). Models of detection based on correlation discrimination are used to predict the relation between the two sets of results. Although results are in general agreement, a small but consistent discrepancy was found in that detection thresholds were always about one decibel better than predicted.

Third, we conducted several studies focused on the interaction of interaural time and intensity. In a study stimulated by our results from unilaterally impaired listeners, we measured the ability to discriminate interaural time delay in conditions with large interaural intensity differences for wide-band noise and for 500-Hz tones (Russell, 1979). We found, consistent with the results from impaired listeners, that interaural intensity differences have small effects with the noise stimulus but large effects with the tone. We hope that the stimulus dependence of this effect will help us to understand the large effect with tones which has been puzzling us for some time. In an attempt to understand the variability that has been observed in many of our time-intensity studies, we measured the effects of variability in the acoustic coupling of earphones to the ear. Specifically, we compared performance in interaural time discrimination studies with and without the use of an acoustic monitor-and-adjust system applied to the signals in the earphone cavities (Giordano, 1979). Even in the regions with the largest behavioral variability, the monitor-and-adjust system does not appear to be a significant factor compared with the internal variability. Another study (Schaefer, 1979), also motivated by reducing variability, compared training and performance for two paradigms: the standard 2I,2AFC and a 4I,2AFC paradigm recommended by Smith (1976). The four-interval paradigm was easier for the subjects in conditions that are unnatural and confusing; the four-interval paradigm also gave less variability in these conditions. In a study that was motivated by differences between the interaural dependences observed in masking and in discrimination and that was also motivated by the possibility of comparisons with impaired listeners, we measured interaural time-discrimination performance with narrow-band noise signals in a wide-band noise background for several interaural conditions of the background noise (Ito, Thompson, and Colburn, 1979). We are comparing discrimination with reinforcing interaural conditions in the masker to results with canceling conditions; no significant differences have been observed in our results. These results are also being analyzed in terms of their agreement and disagreement with available models. In a study of image movements for small changes in time delay as a function of reference conditions, we found (Moss, 1979) very complex perceptions and observations that contradicted predictions of our position model (Stern and Colburn, 1978).

Finally, we are studying the process of auditory localization: the acoustic waveforms in the ear canals in response to a click are measured as a function of source position in an anechoic room, and using these results, we attempt to simulate the position of a noise stimulus by generating the expected waveforms in the ear canals using earphones. Even though the waveforms and acoustic signals are being processed very carefully, discrepancies are apparent between performance with real and simulated sources in position-identification experiments (over the surface of a half sphere).

In the experimental studies of impaired hearing, we have also pursued several projects.

We have completed our review of the literature on binaural interaction in impaired listeners (Durlach, Thompson, and Colburn, 1980).

We also conducted experimental studies of subjects with impaired hearing. First, we have completed the study of localization discrimination that was conducted in collaboration with Dr. Rudolf Hausler (Hausler et al., 1979). By measuring discrimination abilities for interaural time and intensity differences and vertical angle as well as horizontal angle at eight reference values around the head, we are able to show consistent patterns of abnormality that correlate with the categories of impairments studied. Among the results of this study are the following: for our subjects with bilateral sensorineural hearing losses, speech-discrimination scores correlated with vertical angle discrimination ability in ways that could not be accounted for by the audiograms; subjects with severe sensorineural losses (75 to 90 dB HL in one subject) were able to discriminate interaural time delay as well or almost as well as many normal listeners when the stimulus was a wide-band noise, although narrow-band performance varied considerably from frequency to frequency and from subject to subject; and performance by subjects with unilateral neurinomas (surgically confirmed vestibular schwannomas with auditory-nerve involvement in every case) was among the worst in each test. We believe that this study will have an important impact on both theories of binaural interaction and our understanding of impaired auditory systems. In the second study, we collaborated on an in-depth study of a single subject with a neurinoma (vestibular schwannoma) involving the auditory nerve on one side (Florentine, Thompson, Colburn, and Durlach, 1979). As part of a large battery of tests, we tested this subject's binaural abilities in interaural time, intensity, and correlation discrimination with a variety of interaural-level differences, including equal SPL, equal SL, equal loudness, and centered. We used both narrow-band and wide-band noises and found no evidence for binaural interaction of any kind (i.e., all abilities could be explained with monaural processing alone). This result is especially significant considering that the subject's audiogram for the affected ear showed a hearing loss of only 20 to 40 dB HL.

Theoretical studies were pursued in several areas. First, theoretical work is included in many of the experimental studies of normal hearing listed above. Second, we are computing the input-output relations of a hypothetical neural network that is designed to process interaural-level differences in a particular frequency region. The computer program has been tested and agrees with analytic computations for special cases that are amenable to analytic treatment. Third, we have begun to develop a theoretical structure for the analysis of binaural interaction abilities in impaired auditory systems (Colburn and Hausler, 1980).

Our work on facilities during the past year has included the design of a two-channel digital-to-analog and analog-to-digital converter system (Jaffe, 1979) that allows output signals to be digitally delayed relative to each other by time delays smaller than the sampling time of the overall system. This converter system when constructed will increase our ability to provide digitally controlled delays with high bandwidth and delay resolution.

Finally, considerable effort has gone into preparing results for publication: four papers were submitted and five more are in draft form and should be submitted within the next two months.

Publications

Colburn, H.S. and R. Hausler, "Note on the Modeling of Binaural Interaction in Impaired Auditory Systems," to appear in G. van den Brink and F.A. Bilsen (Eds.), Proceedings of International Symposium on Psychophysical, Physiological, and Behavioral Studies in Hearing (Delft University Press, Delft, The Netherlands, 1980).

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3. HEARING AID RESEARCH

National Institutes of Health (Grant 5 R01 NS12846)

Louis D. Braida, Francine R. Chen, Michael C. Coln, Steven V. DeGennaro, Nathaniel I. Durlach, Dennis M. Freeman, Bruce L. Hicks, Paul Milner, Patrick M. Peterson, Michael A. Picheny, Charlotte M. Reed, Kenneth I. Schultz, Orlando Sotomayor-Diaz, Edgar Villchur

This research continues to be 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 linear amplification, amplitude compression, and frequency lowering, and have initiated research on the effects of speaking clearly to improve communication with the hearing-impaired.

Research on linear amplification is primarily concerned with evaluating the extent to which Articulation Theory[1] can be used to predict the dependence of speec intelligibility on frequency-gain characteristic and presentation level for listeners with sensorineural impairments. Initial work,[2] based on the assumption that the impairment could be modelled as an additive noise sufficient to account for the threshold elevation, indicated that the performance of listeners with steeply sloping high-frequency losses using amplification systems with high-frequency emphasis was accurately predicted by the theory. In addition, it appears that good theoretical predictions of the optimum frequency-gain characteristic can be obtained for such listeners. Additional insight into these results has been provided by a recent study[3] which compared the performance of impaired listeners to that for normal listeners with losses simulated by masking noise. According to the results of this study, the masking noise simulated many details of the speech reception performance of listeners with high-frequency losses with good accuracy, but only simulated the average scores for listeners with flat losses. To gain more insight into this problem, we have initiated a study of the intelligibility of speech processed by high-, low-, and bandpass filtering for subjects with a variety of audiometric configurations induced by sensorineural hearing loss or simulated by masking noise.

Research on amplitude compression for listeners with reduced dynamic range has continued to focus on syllabic compression and has been primarily concerned with analyzing our initial negative results on multiband compression.[4] During the past year effort has concentrated on the development of a new amplitude-compression system and on the study of the role of training in the perception of compressed speech.

i. A new hardware implementation of the 16-band syllabic compression system has been completed.[5] This system continues to be based on 1/3-octave bands, but incorporates log amplifiers in each band to provide greater input dynamic range, finer amplitude resolution, and more precise gain control. Since this system is based on a dedicated microcomputer, it is capable of functioning both as a compressor and as a speech-level analyzer. Systematic measurements of speech-level distributions are being made to permit the characteristics of compression systems to be established accurately and to analyze the properties of compressed speech.

ii. A new study of the intelligibility of amplitude-compressed speech has been conducted.[6] Two impaired listeners were trained for an extensive period to identify a large set of CV syllables processed by multiband syllabic compression and linear amplification. Results for both listeners indicated substantial training effects persisting for roughly 5000 presentations for the listener with a moderate sloping hearing loss, and roughly 20000 trials for the listener with a severe flat loss. Performance after training indicated no advantage for compression for the listener with the moderate loss, and a small but significant improvement for the severely impaired listener. Additional studies of syllabic compression using listeners with severely reduced dynamic ranges are planned.

Research on frequency lowering for listeners with negligible hearing at high frequencies continued to focus on pitch-synchronous time-dilation techniques that can incorporate warping of short-term spectra. A study of the effect of pitch transformation on the intelligibility of synthetic vowels with lowered formants has been conducted using normal listeners with simulated high-frequency hearing losses.[7] Vowels were synthesized with formant frequencies scaled down by a factor of three from values typical of male speech and with fundamental frequencies either appropriate for the speech or reduced by a factor of two. Results indicate that after training both intelligibility scores and the detailed confusion patterns are unaffected by fundamental frequency reductions up to a factor of two. This suggests that it may be useful to consider including fundamental frequency transformations in future frequency-lowering systems.

Research on the effect of speaking clearly on the intelligibility of speech for impaired listeners has been initiated in order to obtain background information for new methods of matching speech to residual auditory function. In addition, the information gained from this work may contribute to the training of individuals who communicate orally with the hearing-impaired, to the development of procedures for accounting for speaker differences in intelligibility testing, and to basic speech science. Preliminary measurements of the intelligibility of clearly and conversationally spoken sentence material have been made for four listeners with sensorineural hearing loss.[8] Substantial differences in intelligibility scores (18 percentage points) associated with speaking mode were obtained by each listener. Measurements made on the speech waveforms indicated that in clear speech the durations of all speech elements and the number of pauses per sentence increased, and vowels more closely approached target values, but consonant-vowel amplitude ratios increased only for selected consonants. Additional studies currently in progress are concerned with establishing the generality of these results and relating the improvements in intelligibility to changes in the acoustic properties of speech elements associated with the attempt to speak clearly.

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4. TACTILE COMMUNICATION OF SPEECH

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Health Sciences Fund

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The goal of this research is to develop tactile speech communication aids for the deaf and deaf-blind. Our research during the past year can be subdivided into three categories: (a) Study of Tadoma, (b) Development of Artificial Display Systems, and (c) Comparisons of Artificial Systems with Tadoma. A brief summary of work in each category is presented in the following paragraphs.

a. Study of Tadoma

In the Tadoma Method of speech communication, the "listener" receives speech by placing a hand on the talker's face and monitoring actions associated with the speech-production process. We are studying this method in order to document the speech-communication performance achievable with the method and to obtain background for the design of artificial systems. Work on this method performed during the past year includes (i) an extensive survey of Tadoma use throughout the United States, (ii) in-depth studies of the capabilities of three deaf-blind experienced Tadoma users, (iii) supplementary, less detailed studies of four additional deaf-blind, experienced Tadoma users, and (iv) studies conducted with normal, comparatively untrained subjects in which loss of sight and hearing is simulated by the use of a blindfold and masking noise.

The purpose of the survey is to determine the criteria used for selection of Tadoma trainees, the degree of success achieved with different types of individuals, the types of training programs employed, and the identity of successful Tadoma users. This survey has involved the development of a questionnaire and the submission of this questionnaire to roughly 1000 institutions, as well as numerous interviews with administrators and teachers of the deaf and deaf-blind. The results of this effort are currently being organized and prepared for publication. Roughly 20 blind-deaf successful Tadoma users have been identified by this survey for use in our experimental work.

The experimental research on these deaf-blind experienced Tadoma users[1-4] has been directed toward gaining further understanding of the capabilities of the single Tadoma user studied in our preliminary work and broadening our data base by the inclusion of additional subjects. This research has included study of consonant and vowel confusions in syllable identification tasks, the effect of contextual information and speaking rate on word and sentence reception, the characteristics of the Tadoma users' own speech, and general linguistic competence. In addition, in order to gain further insight into the identity and nature of the perceptual cues employed by the Tadoma user, speech-reception experiments have been performed using restricted hand positions.

Although many of the experiments have produced relatively consistent results across subjects (e.g., tests concerned with the identification of consonants or with the reception of sentences spoken at low rates), some of the experiments have led to results that are strongly subject-dependent (e.g., tests concerned with the use of contextual information, with reception capability at high speaking rates, or with linguistic competence).

In general, it is clear from the results of our experiments with these deaf-blind subjects that it is possible to achieve relatively good speech reception via the tactile sense and to develop substantial speech-production capabilities and linguistic competence, even when the loss of sight and hearing occurs at a very early age (e.g., under 2 years). Furthermore, these experiments have led to considerable insight concerning the perceptual cues used by subjects to discriminate and identify various speech segments. The principal task now facing us is to determine why the results obtained on these subjects with this method are so superior to past results obtained with artificial devices reported in the literature. Among the factors that are being considered here are (i) the overall richness and multidimensionality of the display, (ii) the direct tie between the display and the speech-production process, and (iii) the very extensive training of the subjects.

The research on relatively untrained normal subjects with simulated blindness and deafness,[3,5,6] which is directed toward answering the above question and providing controlled comparisons between different tactile speech-communication systems, has focused on discrimination and identification of nonsense syllables, comprehension of sentences (constructed from a very limited, previously learned set of isolated words), and the development of efficient training procedures. Roughly speaking, our preliminary results with these subjects indicate that only relatively small amounts of training are required to enable inexperienced subjects to discriminate nonsense syllables with an accuracy comparable to that achieved by the experienced Tadoma users and that the most difficult task in learning Tadoma for these subjects concerns the comprehension of running speech. Even for this task, however, we have found no evidence that learning Tadoma is more difficult than learning a foreign language. A project is now being initiated in which six normal subjects will spend roughly 5-10 hours/week for a duration of 1-2 years attempting to learn Tadoma.

b. Development of Artificial Display Systems

In addition to studying the performance achieved with the Tadoma method, considerable effort has been directed toward the development of artificial encoding and display systems. Until recently, the only such systems considered have been those that employed a frequency-to-place encoding scheme (i.e., decomposing the speech signal into spectral bands and presenting different frequency bands to different locations on the skin) and a display based on the transducer portion of the optacon.[7,8]

One project is concerned with the development of a new stimulator array.[9] Although the optacon has been used for preliminary experiments, it has many important limitations (i.e., it can only be applied to the finger, the vibrations cannot be controlled in amplitude on an individual basis, etc.). In order to provide increased flexibility and to help us separate out limitations imposed by the particular encoding scheme from limitations imposed by the particular transducer system, we have constructed a new computer-controlled vibrotactile array. This array, which is modeled after an array developed at Princeton,[10] consists of a rectangular configuration of 81-256 vibratory stimulators (piezoelectric bimorph benders) driven synchronously by linear high-voltage transistor-amplifiers under computer control. As in the Princeton system, the array of stimulators is mounted in such a way that it can conform readily to irregular skin surfaces and can be clamped rigidly to maintain a given conformity. Also, stimulator density can be easily changed by substituting sets of bimorphs within the array. Patterns on the display are created on a frame-by-frame basis by specifying the amplitude of excitation applied to each vibrator on an individual basis. Also, excitation wave shape and frame duration can be specified for each frame. To achieve this degree of flexibility, it was necessary to provide a high degree of hardware parallelism (256 digitally programmable high-voltage amplifiers for driving the bimorphs), as well as a sophisticated control system for the high data rates required (10^5 bytes/sec) to specify complex patterns at peak frame rates.

A second project is concerned with the creation of software for the study of different types of encoding and display schemes. For example, we have developed an algorithm for displaying acoustic frequency in terms of a textural variable (spatial frequency) rather than a spatial variable. Similarly, we have developed a linear-prediction scheme to convert acoustical waveforms to representations of vocal-tract area functions (plus voicing detection) for display on our stimulator arrays. Encoding by use of the vocal-tract area function is of interest both because of the low bit rate associated with this type of system (as compared to a channel vocoder) and because of the direct tie to the speech-articulation process.

A third project is concerned with the development of a synthetic Tadoma system. In this system, the listener places his hand on an artificial face that is driven by a sensor array (via computer control) mounted on the talker's face. The initial goal is to create a system that will produce speech reception results similar to those produced with "real" Tadoma. Once this goal has been achieved, the system will be used to gain further insight into Tadoma by altering different components of the system and examining the effects of these alterations on speech reception (i.e., altering the inputs, the processing, and/or the display). Initial work on this project has focused on exploration of techniques for construction of the artificial face (the most difficult task in the project). In this work, we have assumed that all the necessary input signals to the display are available from the sensor array and that the problem is to make use of these signals to drive the artificial face in an appropriate manner. Furthermore, we have assumed that the main problem in the construction of the face concerns the accurate reproduction of lip movement, since we believe that the reproduction of laryngeal vibration, breath flow, and (if necessary) jaw movement can be achieved using relatively standard devices. In the design of the lip-movement system now under consideration, a rubberlike material is used to mold the "lips" and surrounding "skin" and to obtain internal cavities of various sizes and shapes. The position and movement of the lips is then controlled by the air pressure in the internal cavities (which vary in shape as the air pressure is varied). Current work on this lip-display system is focused on the selection of material, the selection of the method for controlling airflow, and the cavity design problem.

Further engineering effort has been directed toward the design of interfaces and software for computer control of the various display systems. For example, we have designed a new interface for the optacon to ensure synchronous generation of display frames, reduce system overhead, and simplify the programming required to specify patterns. We have also developed a general-purpose enhancement to standard DMA controllers to permit real-time clock-controlled input and output transfers to occur with minimal computer loading. This interface design will be used for transfers to the new tactile array system (which requires only output transfers) as well as to the artificial face system (which will ultimately require both input and output transfers). In addition, we have begun to develop the control subsystem of the new tactile array system. This hardware will translate computer-generated specifications of tactile stimulus patterns into timing and data-flow signals that can be applied to the DAC's and amplifiers of the display.

Finally, we are beginning to formulate a program of psychophysical experiments concerned with the determination of appropriate schemes for displaying time-varying functions to the tactile sense.[11] Independent of exactly how these functions are derived from the speech signal (e.g., by taking amplitude spectra, vocal-tract area functions, etc.), the problem of how to best display these functions is clearly fundamental. The above-described engineering efforts concerned with the development of the new vibrotactile array and the various computer programs and interfaces provide us with a facility that is highly suitable for this purpose.

c. Comparison of Artificial Systems with Tadoma

Although the overall speech-reception results obtained with deaf-blind subjects and the Tadoma Method appear to be substantially superior to results obtained with artificial systems, one cannot automatically conclude that the Tadoma Method is superior. In particular, such a conclusion ignores the vast differences in training received by the Tadoma users and the subjects tested with the artificial systems. Clearly, a rigorous comparison requires the use of equivalent subjects and equivalent training.

In a preliminary experiment comparing a frequency-amplitude display on the optacon to Tadoma,[12] 32 pairs of consonants (contrasting voicing, manner, and place) were used in a CV syllable discrimination test with normal, relatively inexperienced subjects. The consonant contrast was held fixed throughout a run, but the vowel was varied randomly (over the set /i,a,u/) from trial-to-trial. All stimuli were presented "live voice" and nontactile cues were eliminated by the use of a blindfold and masking noise. The results of these discrimination tests showed Tadoma to be substantially superior with respect to contrasts of voicing and place, but roughly equivalent to the artificial system for contrasts of manner.

The fact that one must be careful in generalizing this result, however, is indicated by a comparison between our Tadoma results and those reported for the MESA.[13] According to our analysis,[3] Tadoma is superior with respect to the identification of consonants, but inferior with respect to the identification of vowels. Furthermore, within the consonant category, although Tadoma produces superior performance for plosives and nasals, it produces roughly comparable performance for fricatives. Overall, we see no evidence that this artificial display is substantially inferior to Tadoma for the transmission of segmental elements.

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5. MUSICAL PITCH

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Adrian J.M. Houtsma

The overall objective of this research is to gain understanding of the auditory processes that underlie musical pitch sensations evoked by complex sounds. Research effort has been focused on three projects.

a. Pitch Perception of Harmonic Tone Complexes

Evaluation of modern pitch theories, using a musical interval recognition paradigm, was continued.[1] Attention was focused on the influence of the intensity relation between the pure-tone components of two-tone complexes on the ability of subjects to track the missing fundamental.[2] Dichotically presented two-tone complexes with interaural intensity differences of 0, ±10, and ±20 dB and frequencies nf_0 and $(n+1)f_0$ were used to play melodic intervals to be identified. Confusion matrices were determined for each intensity relation and several values of n. Two of the theories studied[3,4] make specific predictions of expected confusions. A third "analytic perception" model, which assumes that the pitch of a complex tone is largely determined by the frequency of the loudest partial, was also tested. Results so far seem to indicate a stronger correlation between the confusion data and the analytic model than between the data and confusions predicted by any of the current theories.

b. Pitch Perception of Amplitude-Modulated Noise

Periodically modulated AM noise has traditionally been regarded as being devoid of spectral clues, and pitch sensations evoked by these sounds have been explained on a temporal basis. Short-term spectra of such signals, however, do contain possibly relevant pitch information. Two models were developed, one based on temporal processing, the other on short-term spectral processing. They were quantitatively tested with experimental data on musical interval recognition using lowpass noise modulated by a sine wave, square wave or a periodic narrow pulse. The empirical data are generally more consistent with the temporal model than with the spectral model. There is, however, some evidence for both types of processing in the auditory system, where temporal processing dominates in the high-frequency channels and short-term spectral processing in the low-frequency channels. Details of this study are available in Houtsma, Wicke, and Ordubadi.[5]

c. Induced Pitch Shifts for Pure and Complex Tones

According to the "virtual pitch theory" by Terhardt,[4] complex tone pitch is derived from spectral pitch images of the partials, rather than from the partial frequencies. It has also been found experimentally that a pure-tone pitch image can be changed by as much as 5% when the tone is partially masked by lowpass noise.[6] Noise-induced shifts in the pure-tone pitches of partials and in the complex-tone pitch of the missing fundamental were studied through pitch matches between partially masked one- and two-tone test stimuli and an adjustable periodic pulse comparison stimulus. Both monotic and dichotic signal presentations were used. The results show a systematic dependence of the pure-tone pitch on the intensity of the masking noise, which is in qualitative and quantitative agreement with other results found in the literature. The complex tone pitch, however, showed a considerably smaller variation with noise level than expected if this pitch were derived from the respective partial (pure-tone) pitches. This finding is inconsistent with the serial processing idea of the virtual pitch theory, but suggests that pure-tone pitch and complex tone pitch are processed via separate parallel channels, the former based on spatial encoding and the latter on temporal encoding. Details are available in Houtsma.[7]

d. Pitch Discrimination of Pure and Complex Tones in Masking Noise

Work has begun on the study of pitch discrimination for pure tones, complex tones, and synthetic speech vowels in the presence of masking noise. A 2-interval forced-choice adaptive procedure is used to measure pitch jnd's as a function of masking noise intensity. The results are used to compare pitch discrimination sensitivity of human observers to that of current pitch extractor algorithms which are used for analysis and synthesis of speech.

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C. Transduction Mechanisms in Hair Cell Organs

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Lawrence S. Frishkopf, Charles M. Oman

Our objective is to understand the mechanisms of transduction in the phylogenetically related auditory, vestibular, and lateral line organs in response to mechanical stimulation. In all of these organs the receptors are ciliated hair cells.

Goals during the past year have been (1) to measure the displacement of the cupula in the skate semicircular canal as a function of applied force, resulting in a description of cupula motion in terms of a simple elastic model; (2) to determine the influence in this system of canal shape on endolymph flow and cupula motion; and (3) to begin studies of mechanical properties of hair cell cilia in a simple auditory organ, the basilar papilla of the alligator lizard.

1. STIFFNESS COEFFICIENT OF THE CUPULA IN THE SEMICIRCULAR CANAL OF THE SKATE*

Lawrence S. Frishkopf, Richard D. Kunin

Cupula displacements have been measured in the semicircular canal of the skate, *Raja erinacea*, under known loads. The cupula stiffness coefficient — the ratio of applied torque to angular displacement — has thereby been directly determined.[1] All procedures were carried out in artificial skate perilymph.[2] The excised labyrinth was dissected to expose the ampulla which was cut from the canal with fine iris scissors. The ampulla was trimmed to enlarge the openings at both ends; contact with the cupula was carefully avoided. Attachment of the cupula at both the crista and at the vault of the ampulla and the integrity of the hair cell cilia were assessed using Nomarski interference contrast optics. Preparations which by these criteria appeared undamaged were studied further under the Nomarski microscope.

Small pieces of aluminum foil were placed upon the cupula and resulting cupula displacements at one or more points were measured. Displacements were in the range from 10 to 160 micrometers, large compared to estimates of the physiological upper limit (3-5 micrometers) in the same species.[3] Displacements at different points were consistent with the notion that the cupula moves as a unit about an attachment region at the crista. The location of the foil allowed estimation of the torque about the crista and thereby determination of the stiffness coefficient of the cupula. In some preparations several pieces of foil were placed upon the cupula and thereby successive increments of displacement were measured; the stiffness coefficient increased as additional load was placed on the cupula. Measured stiffness coefficients were between 4×10^{-4} and 4×10^{-3} dyne-cm/rad in different preparations. These values are consistent with those that have been inferred in other species by less direct means.[4-6]

Footnotes and References

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2. THE INFLUENCE OF AMPULLA, DUCT, AND UTRICULAR SHAPE ON SEMICIRCULAR CANAL ENDOLYMPH FLOW DYNAMICS

Charles M. Oman, Edward N. Marcus

Theoretical descriptions of endolymph flow and cupula motion in the semicircular canal have traditionally been made employing a simple model in which the entire membranous canal is represented as a hollow, thin torus. The canal lumen is thus assumed to be of constant, circular cross section. The traditional model (van Egmond et al., 1949) predicts that angular flow displacement about the center of the torus is related to head angular acceleration by a second-order differential equation with two coefficients. However, the size of the lumen of an actual canal does vary significantly in the ampulla and utricle. In the past, numerous authors have proposed — with no detailed theoretical justification — that the two-coefficient van Egmond/Groen model may be adjusted for the presence of the ampulla and utricle simply by neglecting the effects of flow drag in these segments (Money et al., 1971; McLaren, 1977) or by increasing the moment of inertia assumed for the fluid ring (Mayne, 1965; Oman and Young, 1972; Curthoys et al., 1977).

Oman (1979), recently employed a model with more complex geometry in which the duct and utricle were represented as separate segments, each having a different (elliptic) cross section and arbitrary length. Oman demonstrated that in this case fluid flow is more accurately described by a second-order differential equation with three coefficients, rather than by the traditional two-coefficient model. It was shown that the short time constant of the canal should be relatively independent of the fraction of the canal torus occupied by the utricle and ampulla, but that the amount of cupula motion produced by head movement should be heavily influenced by this factor. Preliminary numerical estimates of semicircular canal short time constant and volume-displacement gain were made using anatomical data available in several species, including man. However, detailed anatomical measurements describing the shape of the ampulla and utricle were not available.

During the past year, we have extended our theoretical modelling approach to describe Newtonian endolymph flow for the yet more general case where the shape of the canal lumen varies continuously in a gradual fashion through the duct, ampulla, and utricle, and where the central axis of the canal lumen lies in a single plane, but may deviate from the commonly assumed circular form. Analytical expressions for the three coefficients of the resulting second-order differential equation for fluid flow have been determined. The coefficients were shown to depend strongly on the average of the inverse cross-sectional area of the canal lumen and the average of the inverse cross-sectional area squared, taken around the entire canal. To investi gate the behavior of these mathematical functions for an actual case, an optical technique was devised which permitted the necessary dimensional measurements to be made in the duct, ampulla, and utricle of a fixed, dissected, horizontal semicircula canal duct of the skate, Raja erinacea. Our numerical results confirm the conclusions reached using Oman's simpler two-segment model. However, the new more general analytical model provides useful theoretical insights. We conclude that, in most cases, the three coefficients of the differential equation describing canal fluid flow are dominated by the influence of narrow-duct cross-sectional area and its shape, and the fraction of the circumference of the canal occupied by the duct.

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XXVII. COMMUNICATIONS BIOPHYSICS

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.