17. Communications Biophysics

A. Signal Transmission in The Auditory System

Academic and Research Staff

Prof. N.Y.S. Kiang, Prof. W.T. Peake, Prof. W.M. Siebert, Prof. T.F. Weiss, Dr. M.C. Brown, Dr. B. Delgutte, Dr. R.A. Eatock, Dr. D.K. Eddington, Dr. J.J. Guinan, Jr., Dr. E.M. Keithley, Dr. J.B. Kobler, Dr. W.M. Rabinowitz, Dr. J.J. Rosowski, J.W. Larrabee, F.J. Stefanov-Wagner, D.A. Steffens

Graduate Students

D.M. Freeman, M.L. Gifford, G. Girzon, R.C. Kidd, M.P. McCue, X-D Pang, C. Rose

17.1 Basic and Clinical Studies of the Auditory System

National Institutes of Health (Grants 2 PO1 NS13126, 5 RO1 NS18682, 5 RO1 NS20322, 1 RO1 NS 20269, and 5 T32 NS 07047)
Symbion, Inc.

Studies of signal transmission in the auditory system continue in cooperation with the Eaton-Peabody Laboratory for Auditory Physiology at the Massachusetts Eye and Ear Infirmary. Our goals are to determine the anatomical structures and physiological mechanisms that underlie vertebrate hearing and to apply that knowledge where possible to clinical problems. This year the Cochlear Implant Laboratory was formed and a joint program was established with the Massachusetts Eye and Ear Infirmary to study subjects with cochlear prostheses. The ultimate goal of these devices is to provide speech communication for the deaf by using electrical stimulation of intracochlear electrodes to elicit patterns of auditory nerve fiber activity that the brain can learn to interpret.

17.1.1 Comparative Aspects of Middle-Ear Transmission

William T. Peake, John J. Rosowski

Our work on comparative aspects of signal transmission through the middle ear aims to relate the structure of the middle ear to its function. This year we focused on the completion of a series of papers on the middle ear of the alligator lizard. Papers have been published on distortion products in the sound pressure in the ear canal[1] and on nonlinear properties of the input impedance at the tympanic membrane.[2] The results suggest that both nonlinear phenomena originate in the inner ear. We began to investigate the extent to which a simple nonlinear model of the inner ear can account for these measurements.

17.1.2 Effects of Middle-Ear Muscle Contraction on Middle-Ear Transmission

Xiao-Dong Pang, William T. Peake

We aim to understand the mechanism through which middle-ear muscle contractions alter signal transmission through the middle ear. Therefore, we have measured changes in acoustic input admittance of the ear and the cochlear microphonic potential in response to sound stimuli during maintained contractions of the stapedius muscle in anesthetized cats.[3] Large changes in middle ear transmission (of as much as 30 dB) occur at low frequencies, and appreciable changes occur up to 10 kHz. For physiologically significant levels of muscle contraction, the stapes is displaced (up to 30 μm) in a posterior direction as expected from the anatomical orientation of the muscle tendon. During this motion the incudo-stapedial joint slides and no motion of the incus is seen. These observations suggest that changes in the impedance of the stapes ligament cause the change in transmission.

17.1.3 Model of the Ear of the Alligator Lizard

Dennis M. Freeman, William T. Peake, John J. Rosowski, Thomas F. Weiss

We have developed a model of the ear of the alligator lizard that relates the sound pressure at the tympanic membrane to the receptor potential of hair cells with free-standing stereocilia. The aim is to understand the signal processing mechanisms in this relatively simple ear, which should help us to understand more complex ears such as those found in mammals. The model contains stages to represent sound transmission through the middle and inner ear, micromechanical properties of the hair-cell stereocilia, mechanoelectric transduction, and the electric properties of hair cells. This model accounts for most of our measurements of hair-cell receptor potentials in response to sound. During the past year we have made substantial progress in preparing this work for publication. Motivated by our initial theoretical studies of stereocilia mechanics, we initiated a further theoretical study of stereociliary motion aimed at assessing the role of fluid properties and of passive mechanical properties of stereociliary tufts in determining tuft motion.[4] Our initial results suggest to us that the inertial forces of fluid origin are appreciable and should not be ignored as they have been in previous studies of stereocilia mechanics.

17.1.4 Relation Between Spike Discharges of Cochlear Neurons and the Receptor Potential of Cochlear Hair Cells

Ruth A. Eatock, Christopher Rose, Thomas F. Weiss

A review of the relation of spike discharges of cochlear neurons and the receptor potential of cochlear hair cells was published.[5] It was concluded that while much qualitative knowledge of this relation is available, quantitative and coordinated studies of these two signals are required to crystallize our understanding of signal processing mechanisms in the cochlea. In this connection, we have investigated tone-induced synchronization of the discharges of cochlear neurons in the alligator lizard. There are two principal results: (1) with increasing frequency the rate of loss of synchronized response of nerve fibers innervating hair cells with free-standing stereocilia exceeds that found in the receptor potential of these hair cells so that synaptic and excitatory mechanisms at the hair-cell neuron junction must contribute to loss of synchronization; (2) the loss of synchronized response of cochlear neurons innervating hair cells with free-standing stereocilia differs from that of neurons innervating hair cells covered by a tectorial membrane. The differences in responses may well be related to distinct differences in synaptic structure and innervation reported for these two groups of neurons.

17.1.5 Accuracy of Algorithms for Neural Iso-Response Measurements

Christopher Rose, Thomas F. Weiss

We investigated the accuracy of automatic algorithms, used in our laboratory and in many others, for measurement of the frequency selectivity of cochlear neurons. We now have a much better understanding of the limitations of such methods. The results should be of practical utility to many investigators.

17.1.6 Coding of Speech in Discharges of Cochlear Nerve Fibers

Bertrand Delgutte, Nelson Y.S. Kiang

We aim to understand the coding of speech in the discharges of cochlear nerve fibers. During this year, we began to examine the role of the mechanisms responsible for two-tone rate suppression on speech coding in cochlear nerve fibers of anesthetized cats. We developed an automatic method to measure iso-suppression contours. Also, we focused on the relation between responses of cochlear nerve fibers and intensity discrimination. We have, in effect, measured an analog of intensity difference limens for single auditory nerve fibers. For stimulus levels within the dynamic range of the fiber, these "single-fiber limens" are commensurate with behavioral difference limens in humans.

17.1.7 Structural and Functional Studies of the Spiral Ganglion

M. Christian Brown, Nelson Y.S. Kiang

The mammalian cochlea contains two types of afferent and two types of efferent neurons. We aim to understand the structure and function of these neurons. During the past year we examined the peripheral terminations of fibers emanating from the spiral ganglion of the guinea pig. Extracellular injections of horseradish peroxidase in the ganglion labelled several types of fibers in the cochlea.[6] Presumed *afferent* fibers, believed to be the peripheral portions of spiral ganglion cells, fell into two classes: radial fibers and outer spiral fibers. Radial fibers, believed to be the peripheral processes of Type I spiral ganglion cells, sent myelinated peripheral fibers to one or two inner hair cells. Outer spiral fibers, believed to be the peripheral processes of type II spiral ganglion cells, sent thin, unmyelinated axons to spiral basally within the outer spiral bundles and innervate several outer hair cells, usually within a single row. Presumed *efferent* fibers coursed in the intraganglionic spiral bundle and innervated the organ of Corti. These fibers fell into two classes, *thick* and *thin* efferents. Thick efferents innervated outer hair cells and could be either unmyelinated or myelinated in the osseous spiral lamina. Thin efferents innervated areas near inner hair cells and had thin, unmyelinated axons. No examples of fibers were found which innervated regions near both inner and outer hair cells. Thus, as with the afferent fibers, there appears to be a separation of efferent fibers according to their terminations within the organ of Corti.

In recordings from the spiral ganglion with glass micropipettes, two physiological classes of units which respond to sound have been observed. One class responds only to sound in the ipsilateral ear, and has sound-induced and spontaneous discharge characteristics that are similar to those of units recorded in the VIIIth cranial nerve. Hence, we assume that these originate from type I afferents. In addition, for three of these units, intracellular injections of horseradish peroxidase have marked type I afferents. A second class of units can be recorded at the peripheral side of the ganglion. These units respond preferentially to sound in either the ipsilateral or the contralateral ear, have responses with long latencies (5 to 100 msec) which depend on sound-pressure level, have either no spontaneous spike discharges or regular spontaneous spike discharges, and have a wide range of characteristic frequencies (0.34 to 14 kHz). We assume these units originate from the thick efferents because of the similarity of their characteristics to identified efferent fibers recorded near the vestibulocochlear anastomosis in the cat.[7] In addition, two such units in the spiral ganglion of the guinea pig (ipsilaterally excited) have been labeled and classified anatomically as thick efferents. To date, no class of units which might correspond to the type II afferent neurons has been observed among the units which respond to sound in the spiral ganglion.

17.1.8 Middle-Ear Muscle Reflex

John J. Guinan, James B. Kobler, Michael P. McCue

We aim to determine the structural and functional basis of the middle-ear muscle reflex. We have labelled stapedius motoneurons following large injections of horseradish peroxidase (HRP) in the stapedius muscle.[8] The most important conclusion of this work is that the stapedius muscle, the smallest muscle in the body, is innervated by an extraordinarily large number of motoneurons (almost 1200 in the cat).

For some muscles that are innervated by multiple nerve fascicles, the central pattern of innervating motoneurons differs for different fascicles. To see if this applies to the stapedius muscle, injections of HRP have been made in single fascicles of the stapedius nerve at the point where the nerve enters the muscle. In each case, retrograde labelling of stapedius motoneurons was found in each of the four major anatomically-defined stapedius motoneuron groups. This result suggests that each of these groups projects relatively uniformly throughout the stapedius muscle.

Manuscripts in preparation describe changes in sound-induced stapedius electromyographic responses following cuts of the facial genu or lesions around the facial motor nucleus. The results show that even though stapedius motoneurons form a single motoneuron pool, their recruitment order is not determined by a single size-principle. Thus, the stapedius motoneuron pool does not fit textbook descriptions of the organization of motoneuron pools.

We have developed techniques for recording from single stapedius motoneurons. We use electrolyte-filled pipets placed in stapedial-nerve fascicles distal to their junction with the facial nerve in cats paralyzed with flaxedil. We have studied approximately 100 stapedius nerve fibers all of which responded to sound with high thresholds, typically 90-100 dB SPL, and broad tuning curves with a minimum threshold in the 1-2 kHz range. Half of the stapedius nerve fibers responded to tones in either or both ears, 15% responded to binaural tones but not to monaural tones (up to 115 dB SPL), 17% responded to ipsilateral tones but not to contralateral tones, and 18% responded to contralateral tones but not to ipsilateral tones. These results are consistent with the electromyographic results and strengthen the conclusion that there are monaural stapedius motoneurons which respond to either ipsilateral or contralateral sound but not both. Therefore the response properties of the stapedius motoneuron pool cannot be described by applying the size principle to the pool as a whole (if the size principle applies at all).

17.1.9 Cochlear Efferent System

Margaret L. Gifford, John J. Guinan, William M. Rabinowitz

There are two groups of olivocochlear neurons that give rise to efferent fibers that innervate the cochlea: medial and lateral olivocochlear neurons. Our aim is to understand the physiological effects of activation of these two groups. We have completed a study of the effects on cochlear nerve fibers of electrical stimulation in the region of the medial olivocochlear neurons.[9]

In order to examine the mechanical effects of stimulating cochlear efferents in the cat, we developed a system for recording small changes in ear-canal acoustic impedance. This impedance depends on the impedance of the tympanic membrane and on the impedance of the cochlea. Electrical stimulation of olivocochlear efferents at the floor of the fourth ventricle changes the ear-canal impedance by as much as 7%. This impedance change is greatest for low-level sounds and decreases for high-level sounds in a manner reminiscent of the effects of efferent stimulation on auditory-nerve discharge rates. The impedance change can be either positive or negative and is a strong function of frequency. Since these impedance changes occur in animals with the middle-ear muscles cut and disappear after the olivocochlear fibers are severed at the lateral edge of the fourth ventricle, they appear to be due to mechanical changes within the cochlea induced by the efferent fibers.

The function of the olivocochlear bundle (OCB) in man is unknown. We have an opportunity to directly examine OCB effects in man by virtue of two circumstances. (1) As the OCB fibers exit the brainstem, they course initially within the vestibular nerve and later they separate to enter the cochlea. (2) To relieve some patients from incapacitating vertigo, they will undergo a unilateral vestibular nerve transection (performed by Dr. J.B. Nadol at the Massachusetts Eye and Ear Infirmary). The procedure should, unavoidably, result in complete unilateral OCB transection and, ideally, no change in afferent auditory function. Such patients are particularly appropriate for the study of OCB effects because the transection is unilateral and the subjects will be available pre- and post-operatively; hence, each subject can serve as his/her own control. We have developed a variety of behavioral and electrophysiological tests designed to elucidate possible OCB effects for use with these patients. Six patients have been tested pre-operatively and four have been tested post-operatively (at 3 months). During the next year, we shall obtain results at 9–12 months post-op for these six patients, and evaluate the results.

17.1.10 Cochlear Implants

Donald K. Eddington, Gary Girzon

During this year we obtained funds for and equipped the Cochlear Implant Laboratory. We also made significant progress on two projects. First, we initiated a project to develop a computer model of the mammalian ear to display the activity patterns of auditory nerve fibers as a function of characteristic frequency and time for arbitrary acoustic waveforms. These "neurograms" will be studied to identify features of the activity patterns that cochlear implant systems should produce for speech recognition. An initial model has been designed and the three dimensional graphics software needed to display the information implemented. Second, our first subject received a six-electrode implant at the Massachusetts Eye and Ear Infirmary and initial testing shows that the relative pitch of the percepts elicited by electrical stimulation of these electrodes is consistent with their tonotopic placement.

References

1. J.J. Rosowski, W.T. Peake, and J.R. White, "Cochlear Nonlinearities Inferred from Two-Tone Distortion Products in the Ear Canal of the Alligator Lizard," Hearing Res. 13, 141-158 (1984).
2. J.J. Rosowski, W.T. Peake, and T.J. Lynch, III, "Acoustic Input-Impedance of the Alligator-Lizard Ear: Nonlinear Features," Hearing Res. 16, 205-223 (1984).
3. X.D. Pang and W.T. Peake, "Effects of Stapedius Muscle Contraction on Stapes and Incus Position and on Acoustic Transmission in the Cat," Abstr. Eighth Midwinter Res. Meeting, Assoc. Res. Otolaryngol., Clearwater, Fla., Feb. 3-7, 1985, 72.
4. T.F. Weiss and D.M. Freeman, "Theoretical Studies of Frequency Selectivity, Tonotopic Organization, and Compressive Nonlinearity of Stereociliary Tuft Motion in the Alligator Lizard Cochlea," Abstr. Eighth Midwinter Res. Meeting, Assoc. Res. Otolaryngol., Clearwater, Fla., Feb. 3-7, 1985, pp. 177-178.
5. T.F. Weiss, "Relation of Receptor Potentials of Cochlear Hair Cells to Spike Discharges of Cochlear Neurons," Ann. Rev. Physiol. 46, 247-259 (1984).
6. M.C. Brown, "Peripheral Projections of Labeled Efferent Nerve Fibers in the Guinea Pig Cochlea: An Anatomical Study," Abstr. Eighth Midwinter Res. Meeting, Assoc. Res. Otolaryngol., Clearwater, Fla., Feb. 3-7, 1985, pp. 9-10.
7. M.C. Liberman and M.C. Brown, "Intracellular Labeling of Olivocochlear Efferents near the Anastomosis of Oort in Cats," Abstr. Eighth Midwinter Res. Meeting, Assoc. Res. Otolaryngol., Clearwater, Fla., Feb. 3-7, 1985, 13-14.
8. M.P. Joseph, J.J. Guinan, Jr., B.C. Fullerton, B.E. Norris, and N.Y.S. Kiang, "Number and Distribution of Stapedius Motoneurons in Cats," J. Comp. Neurol. 232, 43-54 (1985).
9. M.L. Gifford, "The Effects of Selective Stimulation of Medial Olivocochlear Neurons on Auditory Nerve Responses to Sound in the Cat," Ph.D. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1984.

B. Auditory Psychophysics and Aids for the Deaf

Academic and Research Staff

L.D. Braida, H.S. Colburn, L.A. Delhorne, N.I. Durlach, K.J. Gabriel, D. Ketten, J. Koehnke, N. Macmillan, W.M. Rabinowitz, C.M. Reed, M.C. Schultz, V.W. Zue, P.M. Zurek

Graduate Students

C.A. Bickley, D.K. Bustamante, C.R. Corbett, M.M. Downs, R. Goldberg, Y. Ito, M. Jain, D.F. Leotta, M.V. McConnell, R.C. Pearsall II, J.C. Pemberton, P.M. Peterson, M.P. Posen, K.A. Silletto, R.C. Thayer, S. Thomson, M.J. Tsuk, R.M. Uchanski

17.2 Binaural Hearing

National Institutes of Health (Grant 5 R01 NS10916)

H. Steven Colburn, Nathaniel I. Durlach, Kaigham J. Gabriel, Manoj Jain, Janet Koehnke, Patrick M. Zurek

Research on binaural hearing continues to involve i) theoretical work, ii) experimental work, iii) development of facilities, and iv) preparation of results for publication.

i) In the neural modeling area, we have developed some simple models for neural structures that could be related to available results from the lateral and medial superior olivary regions. During the past year, we have simulated the behavior of the model neurons and found substantial consistency with the physiological data.

We have also continued to develop our psychophysical model. This model, which is a refinement of previous models discussed in the literature, is being applied to results from both normal and hearing-impaired listeners with encouraging results. The binaural component of the model postulates the following operations: peripheral critical-band filtering, estimation of the interaural time difference as a function of time, estimation of the interaural intensity difference as a function of time, a combination of these estimates through interaural time-intensity trading, and temporal averaging of the composite interaural cue. (The temporal averaging is a critical element for the ability to predict interaural time and intensity discrimination and binaural detection with a common model.)

Further theoretical work has been concerned with the development of a quantitative model of the effects of a conductive hearing loss on binaural interaction. This model proposes that bone-conducted signals are abnormally large relative to air-conducted components in cases of severe conductive impairment, resulting in a loss of cochlear isolation. Predictions of this model for interaural discriminations are made after specifying two parameters (cross-over attenuation and delay). This model predicts one of the more curious results of our experimental study — that interaural time jnd's are enlarged by conductive loss but interaural amplitude jnd's are not. A manuscript on this topic is being prepared for publication (Zurek, 1985).

ii) We have completed three experimental studies of binaural interaction. The first is a study of the dependence of the masking level difference on bandwidth. In addition to a complete set of NoSo and NoSpi thresholds at 250 and 4000 Hz as a function of masker bandwidth, interaural amplitude and phase jnd's and homophasic amplitude jnd's have been measured. The jnd's allow us to test our current model of binaural interaction against this broad data set. A manuscript describing this work is nearing completion (Zurek et al., 1985).

The second is a study of interaural correlation and binaural detection at fixed bandwidth (Koehnke et al., 1985). Psychometric functions for NoSo, NoSpi, interaural correlation discrimination, interaural time discrimination, interaural intensity discrimination, and monaural intensity discrimination were measured for four normal subjects. Results demonstrate, among other things, that NoSpi detection and interaural correlation discrimination are closely related: when correlation is expressed as the equivalent signal-to-noise ratio, the psychometric functions for the two tasks are essentially identical.

The third experimental study, again concerned with detection and interaural correlation discrimination (Jain et al., 1985), focused on issues related to the differences in bandwidth dependence between the two tasks. Specifically, we measured a set of cases that included interaural correlation discrimination for a narrowband noise stimulus in the presence of a spectral fringe of noise with fixed correlation. The results of this experiment, also consistent with the model, show that wideband detection and fringe correlation are compatible in the sense described in the preceding paragraph.

Additional experimental studies were undertaken to examine the feasibility of achieving supernormal binaural hearing (in localization and detection) through new types of signal processing. Specifically, we have initiated studies of signal processing to simulate a greater headwidth (by magnifying interaural differences) and also to slow-down the rapid interaural fluctuations that occur at high frequencies in detection (by employing a comb filter to transform critical band noise to narrowband noise at these frequencies). Further work on these techniques is planned for the coming year.

iii) Work on laboratory facilities has included further development of sophisticated D/A equipment for the main computer in the laboratory (VAX 11/750) and implementation of a waveform generation package on the stand-alone LSI-11/23 system. The waveform-generation package allows the generation of waveforms of arbitrary spectral content, phase relation and correlation.

iv) The following articles have been submitted for publication: Colburn et al., 1985; Gabriel, 1985; Gabriel and Durlach, 1985; Koehnke et al., 1985; Siegel and Colburn, 1985; Stern and Colburn, 1985; Zurek, 1985 a,b,c; Zurek et al., 1985.

Publications

Colburn, H.S., P.M. Zurek, and N.I. Durlach, in W. Yost and G. Gourevitch (Eds.), "Directional Hearing in Relation to Hearing Impairments and Aids," Chapter in Directional Hearing, (1985), accepted for publication.

Gabriel, K.J. and N.I. Durlach, "Interaural Correlation Discrimination. II. Relation to Binaural Unmasking," submitted for publication.

Gabriel, K.J. "Sequential Stopping Rules for Psychophysical Experiments," J. Acoust. Soc. Am., submitted for publication.

Jain, M., J.R. Gallagher, and H.S. Colburn. "Interaural Correlation Discrimination in the Presence of a Spectral Fringe," in preparation.

Koehnke, J., H.S. Colburn, and N.I. Durlach, "Performance in Several Binaural-Interaction Experiments," J. Acoust. Soc. Am., submitted for publication.

Siegel, R.A. and H.S. Colburn, "Binaural Detection with Reproducible Noise Maskers," J. Acoust. Soc. Am., submitted for publication.

Stern, R.M., Jr. and H.S. Colburn, "Lateral-Position Based Models of Interaural Discrimination," J. Acoust. Soc. Am., accepted for publication.

Zurek, P.M. "The Precedence Effect," in W. Yost and G. Gourevitch (Eds.), Chapter in Directional Hearing, accepted for publication.

Zurek, P.M. "A Predictive Model for Binaural Advantages and Directional Effects in Speech Intelligibility," J. Acoust. Soc. Am., submitted for publication.

Zurek, P.M. "Analysis of the Consequences of Conductive Hearing Impairment for Binaural Hearing", in preparation.

Zurek, P.M., N.I. Durlach, K.J. Gabriel, and H.S. Colburn, "Masker-Bandwidth Dependence in Homophasic and Antiphasic Detection," in preparation.

17.3 Discrimination of Spectral Shape

National Institutes of Health (Grant 1 R01 NS16917)

Louis D. Braida, Lorraine A. Delhorne, Nathaniel I. Durlach, Yoshiko Ito, Charlotte M. Reed, Patrick M. Zurek

This research involves experimental and theoretical work on the ability of listeners with normal and impaired hearing to discriminate broadband, continuous, speechlike spectra on the basis of spectral shape. The stimuli consist of Gaussian-noise simulations of steady state unvoiced fricatives (/f, sh, s/) and the burst portion of unvoiced plosives (/p, t, k/), partially masked by a Gaussian-noise simulation of cafeteria-voice babble. The basic paradigm employed is symmetric, two-interval, forced-choice, discrimination with trial-by-trial feedback. To prevent listeners from basing judgements on loudness cues, the overall level of the stimulus is roved between presentation intervals. The data are then summarized in terms of psychometric functions showing the dependence of sensitivity (d') on signal-to-babble ratio (S/B).

Data have been collected on normal listeners for the three pairs of fricatives and the three pairs of plosives, for various roving-level ranges, and for various stimulus durations. Overall sensitivity is essentially unaffected by roving-level: the average d' is roughly the same for a 60 dB rove and a 0 dB rove. Furthermore, the dependence of sensitivity and response bias on level-pair within the 60 dB rove is minimal. In general, the psychometric functions are very shallow: their slopes are in the range 0.1-0.2 d'/dB. Also, the dependence on duration shows strong integration effects: the value of S/B corresponding to d' = 1 changes by roughly 20 dB as the duration changes from 10 msec to 300 msec.

Data have also been collected on two listeners with sensorineural impairments. The results for these listeners fall within the range of results obtained on the normals with one major exception: performance is exceptionally poor when a long stimulus duration is combined with a large rove. In other words, at long stimulus durations (but not short durations), roving-level has a strong abnormal negative effect. Alternately, when the level is roved (but not when it is held fixed), performance at long durations fails to show the normal improvements over performance at short durations (i.e., the normal integration effect does not appear). Auxiliary experiments are now being performed to explore these results further.

Theoretical work has focused on the computation of optimum performance for roving-level spectral-shape discrimination[1] on models of central processing for discrimination of broadband signals, and on the application of a specific black-box model of spectral-shape discrimination to the above-mentioned data and to data in the literature.

The model that is currently being applied to the data includes a peripheral processor that derives an internal representation of the stimulus spectrum and a central processor that relates the internal spectrum to the performance of the listener. The peripheral processor consists of a bank of N filters, a level estimator that squares and integrates each filter output, and a logarithmic transformer. Each level estimate is assumed to be corrupted by an additive internal noise. The central processor consists of an ideal processor that operates on the N-dimensional random vector whose components are the corrupted level estimates. It is assumed that the N additive internal noises are independent identically distributed zero-mean Gaussian random variables. For a given choice of filters and integration times, the model has one free parameter: the variance (K) of the Gaussian noise. We have previously reported (RLE Progress Report No. 126) that predictions of the model were in qualitative agreement with our own data on spectral-shape discrimination, although the best value of K for each prediction varied. We recently applied the model to published data on critical bandwidths,[2] intensity discrimination,[3] and frequency discrimination.[4] The predictions were in qualitative but not quantitative agreement with the data. Current work is concerned with incorporating into this model improved estimates of internal filters and integration times. We are also studying the effects of correlation among the noise components in different frequency channels.

References

1. L.D. Braida, Y. Ito, C.M. Reed, N.I. Durlach, "Optimum Central Processor for Spectral-Shape Discrimination", J. Acoust. Soc. Am., Suppl. 1, 77, S80 (1985).
2. C.M. Reed and R.C. Bilger, "A Comparative Study of S/N and E/N," J. Acoust. Soc. Am. 53, 1039–1044 (1973).
3. W. Jesteadt, C.C. Wier, and D.M. Green, "Intensity Discrimination as a Function of Frequency and Sensation Level," J. Acoust. Soc. Am. 61, 169–177 (1985).
4. C.C. Wier, W. Jesteadt, and D.M. Green, "Frequency Discrimination as a Function of Frequency and Sensation Level," J. Acoust. Soc. Am. 61, 178–184 (1977).

17.4 Role of Anchors in Perception

National Science Foundation (Grants BNS83-19874 and BNS83-19887)

Louis D. Braida, Nathaniel I. Durlach, Rina Goldberg, Neil A. Macmillan, Michael J. Tsuk

This research seeks to provide a unified theory for identification and discrimination of stimuli that are perceptually one-dimensional. During this period, we have i) begun to extend our existing theory, developed for intensity perception, to other continua, ii) conducted experiments using two such continua, and iii) devised psychophysical methods appropriate for large groups of experimental subjects, each of whom serves only briefly.

i) Our current theory of intensity perception[1] postulates *perceptual anchors* near the extremes of the stimulus range. Observers judge stimuli by measuring the perceptual distance from a sensation to the anchors. The model accounts for the increase in sensitivity observed near the edges of the range in intensity identification.

For many other continua, especially those which are "categorically" perceived, sensitivity reaches a peak in the interior of the stimulus range. If such peaks reflect regions of natural high sensitivity, any resolution task including fixed discrimination is predicted to show a peak. If instead they reflect central anchors, identification and roving discrimination are predicted to show sharper peaks than fixed discrimination. Macmillan[2,3] has demonstrated that some peaks reported in the literature are of each type. We are currently developing a quantitative version of the model for continua with central anchors.

ii) We have applied the anchor model to a continuum of synthetic, steady-state vowels[4] in the range /i – I – ɛ/. Resolution was measured in four discrimination conditions (two-interval forced-choice and same-different, fixed and roving discrimination), and in identification conditions with and without a standard. In identification and roving discrimination, resolution was poorer than in fixed discrimination, but not uniformly so: sensitivity differences among tasks were smallest near category boundaries. Presentation of a standard improved performance near the standard, especially in the /i/ region, but lowered resolution for stimuli far from the standard. These results are qualitatively consistent with the anchor model.

Tsuk[5] studied identification of lateralized noise bursts with and without standards. Presence of a standard improved performance, but not in the region of the standard.

iii) In this research, the amount of data gathered in a specific condition on one day is often small; sensitivity estimates are obtained by pooling data across days or subjects. In future work, we plan to study the way in which anchors develop, and the need to pool data will be even more severe. We have therefore compared the statistical properties of sensitivity estimates (d') obtained by pooling data with those obtained by averaging sensitivities. Although pooling data introduces a slight bias, it is statistically more efficient than averaging. A paper reporting this work has been accepted for publication.[6]

References

1. L.D. Braida, J.S. Lim, J.E. Berliner, N.I. Durlach, W.M. Rabinowitz, and S.R. Purks, "Intensity Perception. XIII. Perceptual Anchor Model of Context-Coding," J. Acoust. Soc. Am. 76, 722-731 (1984).
2. N.A. Macmillan, "Memory Modes and Perceptual Anchors in Categorical Perception," Paper presented at Acoust. Soc. Am., Norfolk, Virginia (1984); abstract in J. Acoust. Soc. Am. 75, Suppl. 1, S66 (1984).
3. N.A. Macmillan, "Beyond the Categorical/Continuous Distinction: A Psychophysical Approach to Processing Modes," in S. Harnad (Ed.), Categorical Perception, accepted for publication.
4. R. Goldberg, N.A. Macmillan, and L.D. Braida, "A Perceptual-Anchor Interpretation of Categorical Perception on a Vowel Continuum," Paper presented at Acoust. Soc. Am., Austin, Texas (1985).
5. M.J. Tsuk, "An Objective Experiment to Measure Lateralization," S.B. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1984.
6. N.A. Macmillan and H.L. Kaplan, "Detection Theory Analysis of Group Data: Estimating Sensitivity from Average Hit and False Alarm Rates," Psychological Bulletin, accepted for publication.

17.5 Hearing Aid Research

National Institutes of Health (Grant 5 R01 NS12846)

Louis D. Braida, Diane K. Bustamante, Lorraine A. Delhorne, Nathaniel I. Durlach, Miles P. Posen, Charlotte M. Reed, Karen Silletto, Rosalie M. Uchanski, Victor W. Zue, Patrick M. Zurek

This research is directed toward improving hearing aids for persons with sensorineural hearing impairments. We intend to develop improved aids and to obtain fundamental understanding of the limitations on such aids. The work includes studies of i) effects of noise on intelligibility, ii) amplitude compression, iii) frequency lowering, and iv) clear speech.

i) This project is concerned with the deleterious effects of noise on speech reception by the hearing-impaired. In particular, we are trying to determine the extent to which this extra difficulty in noise reflects aspects of hearing impairment beyond loss of sensitivity. To obtain insight into this problem we have measured monaural speech intelligibility for CV syllables presented in a background of Gaussian noise with the spectral shape of babble. Speech and noise are added prior to spectral shaping with either a flat or a rising frequency-gain characteristic. Over the past year our sample has been increased to seventeen ears of twelve hearing-impaired subjects with roughly 30-60 dB losses and a wide variety of audiometric configurations. The results have been analyzed in terms of a model that assumes that the hearing loss is caused by an equivalent additive noise[1] and Articulation Theory is used to predict intelligibility. Plots of consonant identification score versus Articulation Index for the conditions tested continue to show a strong degree of overlap. If conditions yielding approximately constant Articulation Index are plotted versus average hearing loss (average at 0.5, 1.0, and 2.0 kHz), intelligibility scores from these conditions and the various subjects are roughly constant, independent of hearing loss.

ii) Many sensorineural hearing impairments are characterized by reduced dynamic range and abnormally rapid growth of loudness. Multiband amplitude compression has been suggested to improve speech reception for listeners with such impairments, but the intelligibility advantages associated with multiband compression are limited by distortion of speech cues associated with the short term spectrum.[2,3] To overcome this problem we are studying 'Principal Component Compression,' a means of inter-band control of compressor action that seems capable of achieving significant reductions in level variation with minimal distortion of the short-term spectrum.[4]

A perceptual study to evaluate the ability of principal component compression to compensate for reduced dynamic range in sensorineural hearing impairments is currently in progress. Two types of principal component compression are being examined: compression of the lowest-order component (PCA), which corresponds roughly to compression of the overall level, and compression of the two lowest-order components (PCB), which corresponds roughly to compression of both overall level and spectral slope. For comparison purposes, wideband compression (WC), independent multiband compression (MBC), and linear amplification (LA) are also being tested. Four subjects with sensorineural hearing impairments are being tested with consonant-vowel-consonant (CVC) nonsense syllables and phonetically balanced (Harvard) sentences at the subject's most comfortable listening level (MCL) and at MCL - 10 dB. Collection of the CVC intelligibility data has been completed; collection of the sentence intelligibility data is in progress. Preliminary analysis of the CVC data (in terms of percent-correct phoneme scores) indicate that at MCL all systems are roughly equivalent for each of the four subjects, but that at MCL-10 dB the WC and PCA systems are superior (by 10-15 percentage points) for three of the four subjects.

A study has been completed on infinite peak clipping as a means of amplitude compression.[5] This extreme form of range reduction was evaluated with respect to: 1) the actual amount of compression achieved when clipping is preceded or followed by filtering and 2) spectral distortion. It was found that the widths of level distributions of clipped and post-filtered speech were relatively independent of the characteristics of both the pre-filter and the post-filter. Measured ranges between the 10% and 90% cumulative levels were about 10–15 dB as compared to the input ranges of 30–40 dB.

The clipped spectra of unvoiced speech sounds can be predicted analytically. The clipped spectra of voiced sounds cannot be easily predicted and so several cases were examined empirically. In general, despite the radical distortion of the input waveform, only moderate spectral distortions were found. Further evaluation of infinite peak clipping as a means of amplitude compression will involve perceptual tests employing impaired listeners.

iii) Frequency-lowering is a form of signal processing intended to make high-frequency speech cues available to persons with high-frequency hearing loss. We have been investigating the intelligibility of artificially coded speech whose spectrum is confined to frequencies below 500 Hz. A study of the intelligibility of coded consonants[6] is now being extended to include vowels and diphthongs. Vowels were generated by varying the spectral shape of a complex of ten pure tones spaced at multiples of 50 Hz in the region 50–500 Hz. The relative amplitudes of the components for each vowel were determined from natural utterances by measuring the amplitude of the spectra at each multiple of 500 Hz between 500 and 5000 Hz and then imposing this amplitude on a pure tone ten times lower in frequency than the original. Identification experiments are being conducted to compare the intelligibility of the coded vowels to that of natural utterances lowpass filtered to 500 Hz. Preliminary results indicate that the intelligibility of the coded vowels (90–95% correct) is somewhat higher than that observed under lowpass filtering of natural speech with one token of each vowel.

In conjunction with this work, we have begun to investigate the effect of the type-token ratio on the perception of speech sounds.[7] Performance on sets of lowpass-filtered CV syllables was studied as a function of the number of tokens per consonant in each syllable set. Twenty-four consonants were represented in all the syllable sets and all tokens were produced by one female talker. The number of tokens per consonant in a given experiment ranged from one (one utterance of each syllable in C/a/ context) to three (three utterances per syllable in C/a/ context) to nine (three utterances per syllable in each of three contexts — C/a/, C/i/, C/u/). As the number of tokens increased from one to three to nine, identification scores decreased from 79 to 64 to 57 percent correct. Thus, the size of the effect appears to be substantial for small numbers of tokens. Further work will be directed towards determining how this effect varies with the particular speaker and utterances employed and towards modeling the observed effects.

iv) The ultimate goal of our research on clear speech is the development of improved signal-processing schemes for hearing aids. Picheny[8] (1981) and Chen[9] (1980) have demonstrated the ability of naturally produced clear speech to increase intelligibility for both normal and hearing impaired listeners. The continuing research efforts are directed at more detailed descriptions of the differences between clear and conversational speech and an understanding of the perceptual implications of these differences.

Since one of the major differences concerns temporal factors, durations of phonemes and pauses are being measured. Conversational phoneme duration, clear phoneme duration, and percentage increase (PI) have been measured for one of three speakers. Several one-way ANOVA's were performed with the following factors: (i) syllable stress, (ii) prepausal lengthening (or word position in a sentence), and (iii) the surrounding phoneme environment.

The values of PI determined for various phonemes can be summarized as follows: {aa,ae,ao,aw,ey,oy,eh,b,d,g,dh} — PI<40%; {er,iy,yu,ah,ih,uh,y,p,t,k,m,n,ng,s,v,z,ch,jh} — 40<PI<80%; and {ay,ow,uw,l,r,w,f,th,sh,h} — PI>80%. Results from the ANOVA tests indicate that, in general, (i) stress is not a statistically significant factor and (ii) prepausal lengthening is statistically significant for many groups of phonemes. Phoneme position at a phrase boundary has a significant effect on duration in both speaking modes and an effect on PI. The prepausal effect accounts for roughly 30% of the total variance in the duration measures. The significance of the prepausal effect implies not only that grammatical structure is important in the production of both conversational and clear speech, but that its effect is different in clear and conversational speech.

References

1. R.L. Dugal, L.D. Braida, and N.I. Durlach, "Implications of Previous Research for the Selection of Frequency-Gain Characteristics," in G.A. Studebaker and I. Hochberg (Eds.), Acoustical Factors Affecting Hearing Aid Performance and Measurement (University Park Press 1980).
2. L.D. Braida, N.I. Durlach, S.V. DeGennaro, P.M. Peterson, and D.K. Bustamante, "Review of Recent Research on Multiband Amplitude Compression for the Hearing Impaired," in G.A. Studebaker and F.H. Bess (Eds.), The Vanderbilt Hearing-Aid Report, Monographs in Contemporary Audiology, 1982.
3. S.V. DeGennaro, L.D. Braida, and N.I. Durlach, "Multichannel Syllabic Compression for Severely Impaired Listeners," Proc. 2nd Int. Conf. on Rehab. Engr., Ottawa, Canada, 1984.
4. D.K. Bustamante and L.D. Braida, "Principal Component Amplitude Compression," J. Acoust. Soc. Am., Suppl. 1, 74, ZZ3, S104 (1983).
5. K.A. Silletto, "Evaluation of Infinite Peak Clipping as a Means of Amplitude Compression," M.S. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1984.
6. K.K. Foss. "Identification Experiment on Low-Frequency Artificial Codes as Representations of Speech," S.B. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1983.
7. S.R. Dubois, "A Study of the Intelligibility of Low-Pass Filtered Speech as a Function of the Syllable Set Type-Token Ratio." Course 6.962 Project Paper, Department of Electrical Engineering and Computer Science, M.I.T., 1984.
8. M.A. Picheny, "Speaking Clearly for the Hard of Hearing," Ph.D. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., June 1981.

9. F.R. Chen, "Acoustic Characteristics and Intelligibility of Clear Conversational Speech at the Segmental Level," unpublished M.S. Thesis, Department of Electrical Engineering and Computer Science, M.I.T.

17.6 Multimicrophone Monaural Aids for the Hearing-Impaired

National Institutes of Health (Grants 1 RO1 NS21322-01 and 5 T32-NS07099-07)

Nathaniel I. Durlach, Darlene Ketten, Michael V. McConnell, Patrick M. Peterson, William M. Rabinowitz, Robert C. Thayer, Patrick M. Zurek

The goal of this work is the development of systems that utilize multiple spatial samples of the acoustic environment and form a single-channel output to provide enhanced speech intelligibility for monaural listening by the hearing-impaired. There are two basic problem areas in this project. The first, to which most of our efforts thus far have been addressed, is the achievement of high directionality with a cosmetically-acceptable body-worn array of microphones. The second, which assumes that such directional channels can be achieved, concerns coding the channels (prior to summing for monaural presentation) so that the listener can attend to any single channel while simultaneously monitoring the other channels.

Work on the directionality problem is proceeding along three lines. In the first, we consider arrays of microphones with fixed linear weights to provide maximal frontal directivity (when placed on the head). Questions of interest include the complexity of the effects of diffraction and scattering from the body and head, reliability of acoustic response with repeated placement of the device, and performance (intelligibility) in non-anechoic environments. In addition to beginning work on the design of devices, we are adapting the Speech Transmission Index[1] to our application in order to predict the effects of background noise, room reverberation, and directionality on intelligibility.[2]

The second class of processing schemes involves adaptive linear arrays, that is, linear processing of the microphone signals with complex weightings that depend on estimates of the acoustic environment. Questions of special interest here concern the interactions of the estimation procedure with the complexity of the environment and the rate of convergence of the adaptive algorithm relative to the rate of change of the acoustic environment. Initial work has included 1) development of a computerized procedure for measuring intelligibility thresholds and 2) an extension of the image method for simulating a room's acoustic response[3] to the case of multiple microphones.[4]

The third class of schemes uses nonlinear processing. These schemes are motivated principally by an understanding of the normal binaural processing of signals that leads to enhanced speech intelligibility.[5] Thus far we have attempted to replicate a processing scheme for reproducing the "cocktail-party effect".[6] Our results, when we use the same method of evaluation, are consistent with their results: interference that is spatially separate from the target is suppressed. However, on direct tests of target intelligibility, the processing shows no improvement. Further work now being performed with this basic scheme includes the development of a multiband system with independent processing in each band.

References

1. H.J.M. Steeneken and T. Houtgast, "A Physical Method for Measuring Speech-Transmission Quality," J. Acoust. Soc. Am. 67, 318-326 (1980).
2. R.C. Thayer, "The Modulation Transfer Function in Room Acoustics," S.B. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1985.
3. J.B. Allen and D.A. Berkley, "Image Method for Efficiently Simulating Small-Room Acoustics," J. Acoust. Soc. Am. 65, 943-950 (1979).
4. P.M. Peterson, "Simulation of the Impulse Response Between a Single Source and Multiple, Closely-Spaced Receivers in a Reverberant Room," J. Acoust. Soc. Amer. 76, S50(A) (1984).
5. P.M. Zurek, "A Predictive Model for Binaural Advantages and Directional Effects in Speech Intelligibility," J. Acoust. Soc. Am., submitted for publication.
6. J.F. Kaiser and E.E. David, "Reproducing the Cocktail Party Effect," J. Acoust. Soc. Amer. 32, 918(A) (1960).

17.7 Tactile Perception of Speech

National Institutes of Health (Grant 1 R01 NS14092-06)

National Science Foundation (Grant BNS77-21751)

Louis D. Braida, Lorraine A. Delhorne, Maralene Downs, Nathaniel I. Durlach, Daniel F. Leotta, Joseph Pemberton, William M. Rabinowitz, Charlotte M. Reed, Scott Thomson

The ultimate goal of this research program is to develop tactile aids for the deaf and deaf-blind that will enable the tactile sense to serve as a substitute for hearing. Among the various components of our research in this area are (i) study of tactile communication methods employed by the deaf-blind, (ii) development of an augmented Tadoma system, and (iii) development of a synthetic Tadoma system.

(i) The primary tactile communication methods employed by the deaf-blind are Tadoma (in which speech is perceived by placing a hand on the face of the talker and monitoring the mechanical actions associated with speech production), tactile fingerspelling, and tactile signing. Research on Tadoma[1] has demonstrated that continuous speech can be transmitted through the tactile sense at speeds and with error rates that are satisfactory for everyday communication (thus producing an empirical "existence theorem" that communication through the tactile sense is possible). Tactile signing and tactile fingerspelling differ from Tadoma in that (a) they are much more widely used than Tadoma, (b) they require special knowledge on the part of the "talker", and (c) they are tactile adaptations of communication methods devised for the visual sense.

Current research is concerned with determining how each of these two methods compares to Tadoma with respect to communication rate and error structure.

Results on tactile fingerspelling indicate that (a) performance is highly consistent across the five subjects tested; (b) although production rates are generally slow compared to the speaking rates used in conjunction with Tadoma, the two methods produce comparable results for sentence reception over a small range of comparable rates and for estimates of rates at which continuous discourse can be tracked (roughly 30 to 40 words per minute); and (c) communication is limited by the maximum rate at which fingerspelling can be produced. The results of an auxiliary study concerned with visual reception of fingerspelled words[2] suggest that intelligibility remains high for rates up to 2–3 times the normal rate of production (achieved through variable-speed playback of videotaped stimuli). Preliminary results from our study of tactile signing show a higher degree of performance variability for American Sign Language (ASL) subjects than for Pidgin Signed English (PSE) subjects. Results available for one ASL subject indicate highly accurate tactile reception of signed sentences at transmission rates roughly equivalent to those achieved in the fingerspelling study.

(ii) Direct observation of Tadoma, plus analysis of Tadoma errors, suggests that performance could be substantially improved by augmenting Tadoma with supplementary information on tongue position. A commercially available electro-palatograph (Rion DP-01) is being used to sense the contact pattern between the tongue and palate and the transducer portion of the Optacon isbeing used to display this tongue-position information. (This system is referred to as the "Palatacon".) The palatograph contains an artificial palate (specified for a given individual) with 63 contact detectors which drive, on a one-to-one basis, a visual display of 63 LEDs arranged in the shape of a palate. A programmable interface (using a DEC LSI-11) has been developed to allow for tabular specification of vibrator action on the Optacon's 24x6 array corresponding to the 63 palatographic contact points. The mapping currently in use preserves the general shape of the palate on the vibratory array. This vibratory array is applied to a finger on the hand not being used for Tadoma.

Preliminary experiments have been performed to measure the ability of two laboratory subjects (with simulated deafness and blindness) using augmented Tadoma to discriminate pairs of stimuli (9 vowel pairs and 9 consonant pairs) that are difficult to discriminate using conventional Tadoma.[3] For consonants, performance on the Palatacon alone was generally superior to that obtained through Tadoma alone and equivalent to that obtained on augmented Tadoma. For vowels, this finding was reversed in that performance through Tadoma alone was superior to that through the Palatacon alone and equivalent to that obtained on augmented Tadoma. Thus, the tactile display of tongue contact with the palate appears to be an effective means of improving consonant discrimination, but not vowel discrimination. Future research will include investigation of alterations in the Palatacon system to improve performance on vowels and investigation of other methods of deriving information on tongue position for vowels.

(iii) In order to explore a variety of research questions that have arisen in the study of Tadoma, we are developing a synthetic Tadoma system.[4] This system is composed of an artificial mechanical face that is driven by computer-controlled signals derived from sensors placed on a talker's face. The facial actions represented (based on our understanding of natural Tadoma) are laryngeal vibration, oral airflow, and jaw and lip movement. Laryngeal vibration was monitored with a small microphone situated on the throat near the larynx; oral airflow was sensed with an anemometer positioned in front of the mouth; and jaw and lip movements were measured in the midsaggital plane using cantilevered strain gauges. The complete array of sensors did not appear to markedly effect speech production. Simultaneous multichannel recordings of these signals have been made[5] from two male and two female talkers for a variety of speech materials representative of those used in our studies of natural Tadoma. These recordings have been digitized (12 bit ADC), edited to separate files for each utterance, and processed to remove known transducer artifacts and to be made appropriate for use with the artificial-face display.[5,6]

In the construction of the artificial face, an anatomical-model skull has been adapted to incorporate the articulatory actions specified above. For jaw movement, a rotational joint was constructed near the condylar process and a drive linkage was connected to a DC servomotor. For lip movements, assemblies were mounted behind the lower and upper teeth with actuators that project (following removal of portions of the central incisors) to the locations of the lips in the midsaggital plane; these actuators attach to the lips (rubber tubing) and are driven via Bowden cables and DC servomotors. Each motor operates within a closed-loop, position-control system with feedback of shaft velocity, via an analog tachometer, and shaft position, via a digital optical encoder. Laryngeal vibration is presented with a bone-conduction vibrator placed in the laryngeal region. Oral airflow is approximated by four flow levels using a pressurized air supply and two high-speed, on/off solenoid valves (the valves output to tubes which deliver jets that exit the center of the mouth opening).

A preliminary evaluation of the synthetic Tadoma system has now been completed.[7] Roving ABX discrimination tests were performed among sets of 15 vowels in CVC contexts and among 24 consonants in CV contexts. Two naive adult subjects, with simulated deafness and blindness, have been tested. Average results for vowel discrimination (68%) were equal to those obtained in a comparable study of natural Tadoma.[8] Initial results for consonant discrimination were lower for the synthetic system (68 vs. 78%) due primarily to poor performance on voicing contrasts (performance on these contrasts was roughly at the chance level of 50%). After modifying the way in which laryngeal vibration was represented on the face, performance on voicing contrasts improved to 80% and the overall consonant score to 74%. While these results with normal subjects are encouraging, further tests of speech recognition with deaf-blind Tadoma users are required to assess the adequacy of the synthetic Tadoma system as a simulation of natural

Tadoma. The achievement of an adequate simulation will necessarily confirm our understanding of Tadoma; more importantly, it will provide a flexible research tool to study how performance is affected by various transformations that cannot be achieved with natural Tadoma.

References

1. C.M. Reed, W.M. Rabinowitz, N.I. Durlach, L.D. Braida, S. Conway-Fithian, and M.C. Schultz, "Research on the Tadoma Method of Speech Communication," J. Acoust. Soc. Am. 77, 247-257 (1985).
2. S. Thomson, "Fingerspelled Stimuli: Accuracy of Reception vs. Transmission Rate," S.B. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1984.
3. J. Pemberton, "Evaluation of Augmented Tadoma as a Method of Communication," S.B. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1984.
5. G.M. Skarda, "An Analysis of Facial Action Signals During Speech Production Appropriate for a Synthetic Tadoma System," S.M. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1983.
6. M.M. Downs, "Processing Facial Action Signals for Presentation to an Artificial Face," S.B. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1984.
7. D.F. Leotta, "Synthetic Tadoma: Tactile Speech Reception from a Computer-Controlled Artificial Face," S.M. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., 1985.
8. C.M. Reed, S.I. Rubin, L.D. Braida, and N.I. Durlach, " Analytic Study of the Tadoma Method: Discrimination Ability of Untrained Observers," J. Speech Hearing Res. 21, 625-637 (1978).

C. Transduction Mechanisms in Hair Cell Organs

Academic and Research Staff

Prof. L.S. Frishkopf, Prof. T.F. Weiss, Dr. D.J. DeRosier[22], Dr. C.M. Oman

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 stereocilia; 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.

17.8 Relation of Anatomical and Neural-Response Characteristics in the Alligator Lizard Cochlea

National Institutes of Health (Grant 5 R01 NS11080)

Lawrence S. Frishkopf

Microscopic observation of the mechanical response characteristics of the auditory papilla (cochlea) of the alligator lizard, *Gerrhonotus multicarinatus*, to acoustic stimulation has revealed that in the basal region of the organ, length-dependent resonant motion of ciliary bundles can account for frequency selectivity and tonotopic organization of basally directed fibers in the auditory nerve.[1-3] In the apical region of the organ, studies of nerve fiber responses indicate that the range of characteristic frequencies is lower and that tuning is sharper than for basal fibers; that, apically, tonotopic organization appears to be absent, as judged by neural studies; and that two-tone suppression, absent in the basal region, is found among apical fibers.[4]

Morphologically, the two regions differ in a number of significant ways:[5] in the apical but not in the basal region hair cells are covered by a tectorial membrane; ciliary bundles are shorter apically (5–10 μm) than basally (12–31 μm); apical hair bundles all have the same morphological orientation, whereas basal bundles do not. The innervation of hair cells in the two regions also differs:[6] apical hair cells alone receive efferent synapses; and whereas some apical fibers send branches along the length of the papilla, basal fibers all terminate locally where they enter the papilla.

A program of experimental research has been proposed that will attempt to test possible relationships among some of these physiological and anatomical characteristics. Particular emphasis will be placed on understanding the basis of frequency selectivity and two-tone

---

[22]Professor, Department of Biology, Brandeis University

interactions observed in apical fibers. Initial experiments will explore the mechanical response of the tectorial membrane and its role in determining ciliary motion.

References

1. L.S. Frishkopf and D.J. DeRosier, "Evidence of Length-Dependent Mechanical Tuning of Hair Cell Stereociliary Bundles in the Alligator Lizard Cochlea: Relation to Frequency Analysis," M.I.T. Research Laboratory of Electronics Progress Report No. 125, 190-192 (1983).
2. L.S. Frishkopf and D.J. DeRosier, "Mechanical Tuning of Free-standing Stereociliary Bundles and Frequency Analysis in the Alligator Lizard Cochlea," Hearing Res. 12, 393-404 (1983).
3. T. Holton and A.J. Hudspeth, "A Micromechanical Contribution to Cochlear Tuning and Tonotopic Organization," Science 222, 508-510 (1983).
4. T.F. Weiss, W.T. Peake, A. Ling, Jr., and T. Holton, "Which Structures Determine Frequency Selectivity and Tonotopic Organization of Vertebrate Cochlear Nerve Fibers? Evidence from the Alligator Lizard," in R.F. Naunton and C. Fernandez (Eds.), Evoked Electrical Activity in the Auditory Nervous System, (Academic Press, New York 1978), pp. 91-112.
5. M.J. Mulroy, "Cochlear Anatomy of the Alligator Lizard," Brain Behav. Evol. 10, 69-87 (1974).
6. M.J. Mulroy and T.G. Oblak, "Cochlear Nerve of the Alligator Lizard," J. Comp. Neurol. 233, 463-472 (1985).