

## 26. Communications Biophysics

### A. Signal Transmission in The Auditory System

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### 26.1 Basic and Clinical Studies of the Auditory System

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Our studies of the auditory system have continued in cooperation with the Eaton–Peabody Laboratory at the Massachusetts Eye and Ear Infirmary. We report projects that were completed during 1982.

Measurements of the input impedance of the cat cochlea are part of our program to evaluate signal transmission through middle–ears and to relate middle–ear function to structural features.<sup>1</sup> We now have rather extensive measurements in one mammalian species (cat) and one reptile (alligator lizard). We plan to extend these techniques to a broader range of vertebrate species to determine the effects of gross structural differences.

Investigations of mechanoelectric transduction in the inner ear continue to be focused on the alligator lizard ear because of its technical and conceptual simplicity.<sup>2,3</sup> These studies have led to a novel hypothesis about the frequency selective mechanisms of the ear in which the receptor cells act not only as mechanoelectric, but also as electromechanical transducers.<sup>4</sup>

Our knowledge of the innervation of mammalian inner–ear receptor cells by cochlear neurons has been unclear, partly because individual neurons could not be conveniently traced over their entire extent. The application of techniques that stain a few cells completely has clearly shown that cochlear neurons are segregated into two classes: the large majority of these neurons have large cell bodies and processes and innervate inner hair cells discretely, i.e., each neuron innervates one or

two inner hair cells; a small fraction of the neurons have smaller cell bodies and processes and innervate outer hair cells diffusely, i.e., each neuron innervates many outer hair cells.<sup>5</sup> Presumably, all of the recordings of fiber activity of the cochlear nerve have been obtained from the larger group; the function of the smaller neurons in the cochlea is unknown.

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## B. Auditory Psychophysics and Aids for the Deaf

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## 26.2 Intensity Perception and Loudness

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This research is concerned with the development of a unified, quantitative theory of intensity perception and loudness, and involves the construction and integration of models of sensory processes, short-term memory, perceptual-context effects, and decision making, as well as psychophysical experimentation. During this period, our work has focused on a new model for context coding based on the use of perceptual anchors.

The new model for context coding accounts for the increased sensitivity observed near the extremes of the range when unidimensional stimuli are identified (the resolution edge effect) by assuming that sensations are estimated relative to noisy perceptual anchors with a stochastic ruler. In previous work we have derived the predictions of the model computationally because we have not been able to determine analytic expressions for the conditional densities of the decision variable. According to these computations, the predicted ROC's are closely approximated by straight lines of unit slope on normal coordinates even though the decision densities are not Gaussian. Thus it should be possible to derive a nonlinear but monotonic transformation of the decision variable such that the conditional probability densities of the transformed variable are (at least) approximately Gaussian. Since these densities must have equal variance if the corresponding ROC's are to have unit slope, the required transformation is simply that which equalizes the variances of the conditional densities of the

decision variable. Moreover, since we have been able to compute moments of the decision variable and thus show that the variance of the decision variable increases parabolically from the edge to the middle of the range, the required transformation is simply the arcsin function used to equate variances in statistics. In further analysis we have shown that the transformed densities have nonzero skewness and consequently can only be approximately Gaussian.

Based on the properties of the transformed variable, we have been able to derive analytic approximations for the sensitivities to be expected in identification experiments which illustrate the dependence of sensitivity on model parameters. According to these predictions, 1) there is an upper bound (determined by the average number of ruler steps) to the total sensitivity which can be achieved by increasing the stimulus range, 2) resolution edge effects (whose magnitude is dependent on anchor position and average number of ruler steps) are to be expected in large-range experiments but not in small-range experiments, 3) for a given stimulus range resolution is independent of the number of stimuli in the set, and 4) when the stimulus range is small, the variance of the decision variable in the context coding mode is the sum of sensation noise and anchor noise, and thus may exceed trace mode variance in fixed-level two-interval discrimination experiments.

We are currently utilizing these analytic predictions to obtain refined estimates of the parameters of the new context-coding model. A manuscript on this model is being prepared for publication.

## 26.3 Hearing Aid Research

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This research is directed toward improving hearing aids for persons with sensorineural hearing impairments. We intend both to develop improved aids and to obtain fundamental understanding of the limitations on such aids. The work includes study of i) Amplitude Compression, ii) Frequency Lowering, and iii) Clear Speech.

i) Research on amplitude compression, directed towards listeners with reduced dynamic range, involves the study of a multiband compression system based on infinite peak clipping, the development of a principal components compression system, and preliminary study of a multiband automatic volume control system.

A syllabic compression system based on infinite peak clipping has been designed, constructed, and preliminarily evaluated psychoacoustically (Hildebrandt, 1982). The device filters the input into three bands each of which encompasses one of the first three formants of speech. Amplitude

compression is accomplished on a band by band basis via infinite peak clipping within each band and refiltering the clipped signals to their original ranges. The three resulting signals are summed to form the output. Tests with listeners with hearing loss simulated by additive noise achieved large intelligibility differences favoring compressed speech over linearly amplified speech that had been frequency equalized. For phonetically-balanced monosyllables scores averaged 4% and 36%, and for words in sentences 17% and 98%, for unprocessed and processed speech, respectively. However, a simple (wide-band) clipper system also showed good performance. These preliminary experiments suggest that peak clipping may have considerable potential as a means of amplitude compression for the severely hearing-impaired. In future work we will attempt to understand the spectral effects of clipping. We will also conduct further intelligibility studies to determine the nature of the perceptual distortions introduced by the processing. Ultimately, we will determine the benefits of clipping schemes for presenting speech to impaired listeners who have very small dynamic ranges.

We have developed a software signal processing system capable of determining and compressing the principal components of the short term speech spectrum. This includes a 16 channel 1/3 octave digital filter bank and a multichannel digital amplitude compression system. The system is also capable of analyzing the long-term characteristics of processed and unprocessed speech. Principal component compression will be first evaluated in terms of its effect on spectral shapes of speech and its effect on speech intelligibility for normal hearing listeners with hearing losses simulated by noise. To gain insight into this novel form of signal processing, a catalog of how the shapes of sample spectra are affected by compression of each of the principal components singly (and of the lower order principal components, in pairs) will be assembled. These data will be used to specify the characteristics of the principal component compression systems for the intelligibility studies on listeners with normal hearing. The results of these experiments will provide a basis for designing of a principal component compression system for hearing impaired listeners.

Automatic volume control (AVC) is a type of nonlinear signal processing that keeps the long-term output level constant over a wide range of input levels. Although widely suggested for use in hearing aids, this application has not been studied rigorously. One limitation on its use is associated with increases in the background noise level when speech components are weak or absent. We have implemented two AVC algorithms on a 16 channel computer controlled amplitude compression system (Coin, 1978). In one, each channel's level is measured and controlled independently. In the other, all channel levels are controlled on the basis of the overall input level. To reduce noise effects, both techniques incorporated a multichannel noise suppression algorithm (McAuley and Malpass, 1980). Both algorithms were found to be capable of reducing the range of speech level variations, and the relative level of noise to speech was decreased as more noise suppression was applied. However, preliminary tests on a listener with normal hearing indicate that both AVC algorithms degraded speech intelligibility in noise. The degradation was exacerbated by the noise suppression algorithm. We intend to continue to study the function of this AVC system and to attempt to remedy its deficiencies. Preliminary analysis suggests that the techniques used to estimate the short term

spectra of speech and noise may be inaccurate and this may be responsible for some of the intelligibility degradations. It also appears that AVC operation may be improved by use of asymmetric attack and decay dynamics. Use of a faster microcomputer to increase responsiveness is also likely to enhance system performance.

ii) Research on frequency lowering, directed towards listeners with poor hearing at high frequencies, has focused on the perception of low-frequency representations of speech achieved by means of synthetic codes.

Further work has been conducted on artificial coding of consonants and vowels with a 500 Hz bandwidth. Modifications of the original consonant code were introduced in an attempt to reduce confusability of particular pairs of sounds (e.g., /m-n/ and /r-w/) and training on identification was conducted with a second subject. After practicing on various subsets of the consonants, her identification score reached 88% on the full set of 24 coded sounds, a result similar to that obtained on the subject tested previously. A set of coded vowels was created using 10-tone complexes of harmonically spaced pure-tone frequencies from 50 to 500 Hz. The relative amplitudes of the components were patterned after those measured for real speech (lowpass filtered to 2500 Hz and linearly compressed by a factor of 5). Amplitude and duration cues were also coded in the vowel set by varying overall amplitude (such that the overall level of half the vowels was roughly 5 dB higher than for the remaining vowels) and duration (such that six of the vowels were 220 msec and four were 150 msec). Identification scores for one subject ranged between 84-88% following a short amount of practice.

Identification performance on the set of 24 coded consonants constructed with a 500 Hz bandwidth limitation is being compared to performance on naturally produced speech lowpass filtered to 500 Hz. A complete set of results is being obtained on three subjects. In conjunction with this project we are investigating the effect of varying the number of natural speech tokens representing each consonant under lowpass filtering.

Future work with these codes will: 1) explore identification of CV, VC, and CVC syllables drawn at random from the set of 24 consonants and 10 vowels that have been constructed so far; 2) examine the effect of bandwidth in constructing artificial low-frequency codes; and 3) explore further alterations of lowpass filtered speech (not necessarily in real time) to enhance intelligibility.

We have also begun to implement a low-frequency coding scheme that is based on modifying lowpass filtered speech and is capable of operation in real time. The first such code we are exploring is one which Lippmann (1980) found to have a ten percentage-point advantage over ordinary lowpass filtering. This scheme involves using the outputs of four high-frequency analysis filters to control the levels of low-frequency noise bands which are added to the original speech signal (lowpass filtered to 800 Hz). We will compare this scheme to lowpass filtered speech in multiple-token identification tests.

iii) Research on clear speech is concerned with the creation of speech materials that are highly intelligible to impaired listeners through the use of techniques that focus on the speech source rather than on signal processing, and with the determination of the acoustical properties of these materials that are responsible for the high intelligibility.

Work on this project has focused on further analysis of the speech materials and intelligibility results obtained by Chen (1980). Naturally produced clear and conversational CV tokens (6 stop consonants and 3 vowels) were analyzed to determine the VOT and the silence duration between the consonant burst and the start of voicing. Analysis of the acoustic timing measurements showed the following significant effects, speaking mode (15% total variance), voicing (48%), and mode x voicing (13%). No significant effects were found for the silent gap (Laves, 1982).

Our attempts to relate acoustic measurements with the intelligibility results have thus far been inconclusive. In one analysis, we attempted to relate listeners' ability to make voiced/unvoiced decisions to known limitations on duration discrimination for tones masked by noise. Although the dependence of the speech results on S/N was roughly correct, analysis of the voicing scores for different speaker tokens with the same VOT revealed that tokens with the same VOT are perceived quite differently. We plan to analyze the spectral characteristics of the stop burst to gain insight into this problem. After we have obtained a complete set of spectral and timing measurements on these materials, we will prepare synthetic CV tokens in which spectral and temporal differences are varied independently, and we will repeat the Chen intelligibility study with these materials.

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## 26.4 Tactile Perception of Speech

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The ultimate goal of our 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 (a) study of tactile communication methods employed by the deaf-blind, (b) development of an augmented Tadoma system, (c) development of a synthetic Tadoma system, and (d) design and evaluation of a wearable, portable aid.

a) We have obtained a substantial set of results on the Tadoma method of communication (in which the deaf-blind person receives speech by placing his hand on the face and neck of the talker and monitoring the various actions associated with the speech-production process). We have recently begun a study of other methods of communication that have evolved within the deaf-blind community itself, including tactile fingerspelling and tactile signing. We are studying these methods to demonstrate that it is possible to communicate language at reasonable rates through the tactile sense and to gain insight into the design of artificial systems.

Our initial goal in the study of tactile fingerspelling and tactile signing is to examine sentence reception as a function of presentation rate and to compare performance of experienced tactile fingerspellers and signers to that measured previously in experienced Tadoma users. Translations of sentence materials used in the study of Tadoma have been made into American Sign Language (ASL) and videotaped by a native deaf signer (who will also administer the tests). Fingerspelled and signed sentence materials will be presented to deaf-blind subjects at various rates of production and subjects will be asked to repeat the sentences either verbally (in the case of fingerspelling) or in sign (for ASL presentation).

b) Based both on direct observation of the Tadoma method and on detailed analysis of the phonemic errors that are made by Tadoma users, we believe that significantly improved results can be achieved by supplementing the information obtained through Tadoma with information on the tongue position of the talker. In order to explore this question empirically, we have developed a system in which information on tongue position is obtained through an electropalatograph (which senses the pattern of contact between the tongue and the roof of the mouth) and presented to the "listener" on a supplementary tactile display derived from the Optacon transducer system.



Experiments are planned with both normal subjects and experienced deaf-blind Tadoma users to determine the extent to which Tadoma performance can be improved by additional information on tongue position, both for segmental and connected speech utterances.

c) To better understand the success of the Tadoma method and to explore transformations of the method that cannot be achieved directly, we are developing a synthetic Tadoma system in which an artificial talking face is driven by signals recorded from the facial actions of a real talking face. During the past year work has continued on acquiring a library of multichannel recordings of facial actions during speech production. Currently, the library includes two repetitions from a single male speaker of a variety of phonemes, words, and sentences drawn from materials used in the testing of Tadoma users. The recordings, originally made on FM analog tape, have been converted (digitized and edited) to computer files appropriate for driving the artificial face.

On the artificial face, a lip-movement system has been integrated into the skull-jaw assembly. The lower lip can move up and down and in and out relative to the jaw, which in turn rotates about an axis through the skull. The upper lip can move in and out relative to the skull. In addition to lip and jaw movements, the face will also be equipped with mechanisms for producing air-flow at the lips and vibration on the neck. When the face is completed, perceptual experiments will be initiated with deaf-blind subjects to determine the adequacy of the system for speech transmission.

d) We are currently designing a wearable, portable aid in which short-term spectral information will be displayed to the abdomen by means of a linear array of vibrators. This system will be field-tested for substantial periods of time by both deaf-blind subjects and sighted deaf subjects. Such long-term field evaluations with wearable aids are essential not only because they constitute the ultimate test of usefulness in "real-world" environments, but also because they provide the only opportunity for serious learning of the new "code".

## 26.5 Discrimination of Spectral Shape by Impaired Listeners

*National Institutes of Health (Grants 1 R01 NS1691701A1 and 5 T32 NS07099)*

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The overall long-term goal of this research is to develop an analytic understanding of auditory perception and speech reception in impaired listeners and to develop improved diagnostic procedures and prosthetic devices for such listeners. The specific goals of the initial phase of this research are to (1) measure the ability of listeners with peripheral auditory loss, as well as listeners with normal hearing and with simulated loss, to discriminate stimuli with speech-like spectra on the basis of spectral shape alone and (2) relate these measurements to underlying auditory abilities by means of psychoacoustic experiments on the same subjects and through the development of

quantitative, theoretical models.

Our interest in the perception of spectral shape is motivated by a number of factors. First, spectral shape constitutes a significant cue in the perception of many speech elements. Second, it plays an important role in the acquisition of environmental information and in the perception of music. Finally, study of spectral-shape perception should prove useful in the attempt to understand auditory perception in general. The degree of complexity inherent in the study of spectral shape provides a useful interpolation between highly simplified, artificial situations usually studied in psychoacoustics and the highly complex, natural situations ordinarily encountered by listeners in everyday life.

Our research focuses on the discrimination of spectral shape in order to confine our attention initially to performance deficits that are likely to be explainable primarily in terms of peripheral auditory deficits. The basic stimuli for these experiments are synthetic versions of steady-state unvoiced fricatives and synthetic versions of the burst portion of unvoiced plosives. These stimuli, in addition to being sufficiently simple to permit interpretation of discrimination deficits in terms of underlying psychoacoustic abilities, also focus on one of the primary speech-perception problems encountered by the hearing-impaired (i.e., the discrimination of consonant place-of-articulation). The discrimination experiments are structured in such a way that the only perceptual cues available to the listener are those related to spectral shape (i.e., timbre). In particular, subjective duration cues are eliminated by equating the physical durations of stimuli and loudness cues are eliminated by randomly varying the intensity of the stimuli to be discriminated.

The experimental work can be divided into three stages, each of which is designed to contribute additional information concerning the abilities underlying spectral-shape discrimination. To the extent possible, the same subjects (both normal and impaired) will participate in all three stages of experimentation. Stage One is concerned with measuring discrimination performance for pairs of synthetic speech spectra as a function of signal duration and signal-to-noise ratio using a roving-level discrimination paradigm. Stage Two experiments are designed to evaluate the listener's ability to discriminate incremental changes in the spectral shape of the synthetic stimuli in various frequency regions. The basic experiment of this stage can be described as a roving-level detection experiment in which the target is a tone and the masker is a synthetic speech-like stimulus. Stage Three is concerned with the interpretation of the results in terms of more basic psychoacoustic factors. Experiments in Stage Three include fixed-level detection of pure tones in the background of a synthetic speech-like stimulus under both wideband and high- and low-pass filtering conditions, in addition to pure-tone frequency and intensity discrimination experiments.

Our initial theoretical work is directed towards a specification of the peripheral-processor component of the model in sufficient detail to allow us, along with our assumption of an ideal central processor, to derive detailed quantitative predictions for the planned experiments. Our general conceptual framework for spectral-shape discrimination assumes that the incoming signal is

analyzed into a number of frequency channels, the level of activity in each channel is estimated, these estimates are normalized relative to each other (to focus on spectral shape rather than overall level or level within a given channel), and this normalized pattern is then used as the internal datum corresponding to the input signal in the decision process. A model will first be constructed to predict results obtained for normal-hearing listeners and then various modifications will be made to allow predictions of reduced performance obtained for hearing-impaired listeners. The following types of difficulties might be expected in cases of peripheral auditory impairment: auditory frequency analysis may be different from normal, estimation of the level of activity within channels may be deficient, or ability to factor out differences in overall level may be reduced.

Experimental work is currently underway on the three stages of experimentation described above with both normal and impaired listeners. A quantitative theoretical model is also being developed to predict results of Stage One experiments for normal subjects.

## C. Transduction Mechanisms in Hair Cell Organs

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

### 26.6 Evidence of Length-Dependent Mechanical Tuning of Hair Cell Stereociliary Bundles in the Alligator Lizard Cochlea: Relation to Frequency Analysis

*National Institutes of Health (Grants 5 R01 NS11080-06 and GM-21189)*

*Lawrence S. Frishkopf, David J. De Rosier*

We have been studying the mechanical responses to sound in a relatively simple auditory organ, the basilar papilla of the alligator lizard.<sup>1,2</sup> Recently the motion of hair cell stereociliary bundles with respect to the underlying basilar papilla has been measured<sup>3,4</sup> as a function of stimulus frequency, hair cell location, and bundle length. Our purpose has been to determine whether such motion provides a basis for frequency selectivity and tonotopic organization observed in nerve fibers to the organ.

The basilar papilla in the alligator lizard is an elongated auditory receptor organ containing hair cells; it is attached to the basilar membrane which vibrates in response to sound. The axis of symmetry of the stereociliary bundle of each hair cell – presumed to be the sensitive direction for displacement<sup>5</sup> – is perpendicular to the long axis of the papilla.

Characteristic frequencies (CF's) of nerve fibers are between 0.2 and 4 kHz and vary systematically with location in the nerve.<sup>6</sup> Yet velocity measurements along the basilar membrane suggest that its frequency response does not change significantly from one region to another and thus does not provide a basis for the observed variation of nerve fiber CF.<sup>7</sup>

Where then does frequency analysis take place? One possibility is that, within the basilar papilla,

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modes of motion may occur which are not evident in the measurements of basilar membrane motion. It has also been suggested that the gradual variation in maximum length of the free-standing stereocilia in the posterior two thirds of the organ, from 30  $\mu\text{m}$  in the 1 kHz region to 12  $\mu\text{m}$  in the 4 kHz region, may provide a micromechanical basis for frequency analysis over this range of CF.<sup>6</sup>

To address these issues, motion of the basilar papilla and of the free-standing hair cell stereocilia that results from acoustic stimulation has been observed under a microscope. The cochlear duct was excised and the basilar papilla with its supporting limbus was mounted in a drop of lizard Ringer's solution across an opening in an air filled chamber; the organ was stimulated with tones (0.2 – 5 kHz; 110 – 140 dB SPL) and viewed stroboscopically at 400 – 800X.

We have previously shown<sup>1,2</sup> that (1) the papilla rocks about an axis parallel to its length in the direction of symmetry of the stereociliary bundles, i.e., in the direction of morphologically-predicted sensitivity of the hair cells; (2) the papilla moves in phase along its entire length; (3) above 3 kHz, the displacement of the high-CF region exceeds that of the rest of the papilla, perhaps providing a partial basis for frequency analysis.

Recent observations<sup>3</sup> show that on stimulation stereociliary bundles are displaced relative to the underlying cuticular surface in the direction of presumed hair cell sensitivity, and that the difference in phase between bundle and cuticular displacements is a function of stimulus frequency and of hair cell position along the basilar papilla. Phase vs. frequency data suggest (1) that stereociliary bundles behave like damped mechanical resonators; (2) that resonant frequency varies inversely with the square of bundle length; and (3) that bundle resonant frequency and neural CF are close in value in corresponding regions of the papilla and nerve. These findings are consistent with an explanation of frequency analysis in the papilla based on length-dependent mechanical tuning of stereociliary bundles.

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