Final Report

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Speech Processors for Auditory Prostheses

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CONTENTS

I. Introduction

The purpose of this project is to design and evaluate speech processors for auditory prostheses. Ideally, the processors will extract (or preserve) from speech those parameters that are essential for intelligibility and then appropriately encode these parameters for electrical stimulation of the auditory nerve. In this final report we will briefly review our efforts to meet these objectives and offer several suggestions for future work.

II. Brief Descriptions of Project Activities

Major activities of our first, two-year project with the Neural Prosthesis Program included the following:

1. Design, build and test a hardware interface to provide a high-bandwidth communications link between an Eclipse computer and implanted electrodes;

2. Develop and apply an integrated field-neuron model of electrical stimulation by intracochlear electrodes;

3. Identify and contrast promising approaches to the design of speech processors for auditory prostheses;

4. Build a computer-based simulator that is capable of rapid and practical emulation of these approaches in software;

5. Develop software for support of basic psychophysical studies and speech testing;

6. Conduct tests with two patients at the University of California at San Francisco (UCSF) to (a) confirm proper operation of the equipment and software indicated in points 1, 4 and 5 above; (b) obtain basic measures of psychophysical performance in one patient fitted with the UCSF transcutaneous transmission system and one patient fitted with the UCSF percutaneous transmission system; and (c) compare the performance of different, multichannel speech processing strategies with the second patient;

7. Design, build and test a portable, real-time speech processor appropriate for use with single-channel auditory prostheses;

8. Develop and apply ensemble models of the spatial and temporal patterns of neural discharge produced by intracochlear electrical stimulation;

9. Help to establish strong collaborations between UCSF, Duke University Medical Center, Washington University Medical Center and RTI, so that parallel series of tests with implant patients can be conducted in San Francisco, Durham and St. Louis;

10. Build laboratory facilities at UCSF and Duke for testing cochlear-implant patients; and

11. Report the results of this project at major conferences and in manuscripts in preparation.

In the following subsections of this section on project activities we will briefly describe the activities indicated in points 1, 2, 4, 6c, 7, 8 and 10 above. Additional information on all project activities can be found in our quarterly progress reports. To facilitate access to this information, Appendix 1 presents a listing of key contents of the quarterly reports.

A. Hardware interface

The hardware interface is designed to provide a safe means of delivering high-bandwidth stimuli to all eight channels of the UCSF electrode array via a percutaneous cable. Outputs on each of the eight channels can be updated under computer control every 50 usec, and the resolution of these outputs is 12 bits. The clock rate of updates can be increased to 60 kHz if the number of simultaneous outputs is reduced to two. The bandpass of each channel is 5 Hz to 60 kHz. In all, then, the "stimulation side" of the hardware interface supports a very high bandwidth of transmission to the electrode array.

In addition to the stimulation circuitry just described, the hardware interface provides a communications link from the implanted electrode array. Specifically, two channels of artifact rejection and analog-to-digital conversion allow measurement of potentials on both stimulated and unstimulated electrodes at each clock pulse. The assignment of the monitored channels is under computer control. The ability to measure potentials at unstimulated electrodes permits studies of electric field patterns produced by intracochlear electrodes and of the shapes and growths of evoked neural potentials in the cochlea. These "intracochlear evoked potentials" may provide objective measures that complement measures we routinely obtain in psychophysical tests with implant patients. Finally, measurement of potentials on stimulated electrodes permits automated determinations of impedances for all electrodes in the array.

B. Computer-based simulator of speech processors

One of the most-important tools developed in this project is our computer-based simulator of speech processors for auditory prostheses. Use of this simulator allows for rapid and flexible emulation of promising coding strategies in software, which in turn allows us to make valid comparisons between many different approaches to processor design in tests with single subjects. In this way, controls are provided for (a) inter-subject differences in pathology (i.e., differences in the densities, stimulus-response properties and loci of surviving neural elements in the cochlea, and possible differences in the integrity of central auditory structures); (b) the type of electrode array used; and (c) apposition of individual monopolar or bipolar-pair electrodes to excitable tissue. These differences among subjects, along with differences in testing procedures among laboratories, have greatly complicated the interpretation of results obtained in previous studies of processing strategies.

The software for the computer-based simulator of speech processors for auditory prostheses includes the following main programs:

CPEXEC -- executive program for managing communications between, and execution of, other programs in the set;

DESIGN -- program for the design of a signal-processing system, in which the user specifies the function and position of each block within a network of blocks;

MODIFY -- program to modify signal-processing systems previously defined by program DESIGN;

PREPARE -- program that transforms files generated by program DESIGN into files that are used by program EXECUTE;

EXECUTE -- program that executes the simulation of signal-processing systems;

SHOWNTELL -- program for display of outputs generated by EXECUTE, either as graphs on the computer console or as acoustic signals produced over the D/A converter;

SAMPLE -- program to sample speech and other data with the A/D converter, and to store these data on disk in contiguous files with identifying headers;

ASNELEC -- program to assign electrode channels to receive data from the outputs of EXECUTE, and to translate these data into the code required for control of and communication with the RTI Patient Stimulator;

TEST -- program to send data out to the electrodes from the files generated by program ASNELEC, and to monitor and log patient responses to processed speech stimuli.

These programs have been in use for over one year and allow for simulation of a wide range of processing strategies. For example, the DESIGN and EXECUTE programs are fully capable of specification and subsequent simulation of every speech processor for auditory prostheses described in the published literature.

C. Integrated field-neuron model

An important element in the design of speech processors for auditory prostheses is knowledge of the "electrical-to-neural transformer" that characterizes the properties of neural responses to stimuli delivered by intracochlear electrodes. These response characteristics are dependent on a variety of factors including (a) the physical locations, dimensions and electrical parameters of the implanted electrodes and (b) the survival patterns and physiological integrity of neural structures in the cochlea. In an effort to understand the complexities of intracochlear electrical stimulation, we have developed an integrated field-neuron model that couples a description of the electric fields produced by intracochlear electrodes with a description of neural responses resulting from the application of such fields. In particular, the field model is used to compute potential gradients (or the profile of voltages) along the path of a neuron and the neural model is used to predict the patterns of responses that are produced with the imposed voltage profiles.

The potential gradients are calculated by an iterative, two-dimensional, finite-difference model of a cochlear cross section, which includes a pair of electrodes in the scala tympani in transverse-plane sections and many pairs of electrodes in spiral-plane sections. Grid points in the model are 20 um apart and resistivities linking the grid points are defined according to published values for resistivities of tissues and fluids appearing in the cross section. The bipolar electrodes are defined as equipotential conductors mounted in an insulating carrier medium. Fixed voltages are assigned to each electrode and the resultant field patterns are computed by iteration for the entire cross section. Finally, the potential levels at points along the loci of cochlear neurons are extracted from the last iteration of the field calculations.

To predict patterns of neural responses to the imposed voltage profiles, a lumped-element model of a myelinated fiber is used. This model is a modification of McNeal's axon model (McNeal, 1976) of resistively-linked, Frankenhauser-Huxley nodes. The modified model includes myelinated axon cable properties and uses mammalian node of Ranvier characteristics instead of the characteristics for Frankenhauser-Huxley frog nodes. Eighteen active nodes are included, each separated by ten myelinated segments. One section includes characteristics of a cell body, resembling the bipolar cells of the cochlea. A system of simultaneous, nonlinear differential equations is solved iteratively to calculate the model's response to any arbitrary stimulus waveform, applied as a voltage profile along the entire length of the axon. The neuron model, in conjunction with the field potential models, constitutes an integrated model of single fiber behavior in the electrically-stimulated cochlea.

Initial applications of the integrated field-neuron model have included (a) simulation of electric field patterns produced in the transverse and spiral planes by the bipolar electrodes of the UCSF array; (b) examination of the sensitivity of placement of such electrodes in the scala tympani in terms of neural thresholds and spread of excitation; (c) examination of the interaction and crosstalk at a single neural element for stimulation of two or more bipolar pairs in the UCSF array; (d) evaluation of likely effects of bone and other impedance boundaries on field patterns produced by the UCSF electrodes; (e) preliminary examination of field patterns produced by other configurations of intracochlear electrodes; and (f) demonstration of basic properties of neural excitation with voltage profiles approximating those produced by intracochlear electrodes. In general, the results of studies a and b above show excellent agreement with the results of _in vivo_ measurements. Also, the results of the remaining studies provide insight into likely properties of stimulation with cochlear implants that have not yet been determined in animal experiments. Detailed presentations of all of these results can be found in our quarterly reports (see Appendix 1).

D. Ensemble models of neural responses evoked by intracochlear electrical stimulation

In addition to the integrated field-neuron model, described in the previous subsection, we have developed a series of ensemble models of the spatial and temporal patterns of neural discharge produced by intracochlear electrical stimulation. Like the field-neuron model, the ensemble models actually consist of two models: one to describe the field patterns generated by intracochlear electrodes and the other to describe the neural responses evoked by the imposed electric fields. The simplest ensemble model couples an exponential-falloff model of field patterns with a mathematical description of strength-duration curves for intracochlear electrical stimulation. This combined model (emsemble model 1) provides a powerful tool for demonstrating basic patterns of neural responses evoked by a wide range of electrodes and stimuli. In addition, it stands on a firm foundation of previous work in which the exponential-falloff model was used (see, e.g., Black and Clark, 1980; Merzenich and White, 1977; O'Leary _et_ al., 1985) and in which measurements of strength-duration relationships were made for various configurations of intracochlear electrodes (Loeb et al., 1983; van den Honert and Stypulkowski, 1984).

As reviewed in detail in our 8th Quarterly Progress Report, results obtained from this simple ensemble model are consistent with the following observations:

1. A parabolic-like profile of latencies is found in the fields of neural responses evoked by stimulation with intracochlear electrodes;

2. The extent of the excitation field, and the shape of the latency profile, are not necessarily constant for stimulation with constant-charge pulses;

3. Fundamental attributes of the auditory stimulus, such as intensity and frequency, might be coded by appropriate manipulations in the latency profiles of evoked neural responses;

4. Severe interactions between channels can be produced by simultaneous stimulation of different electrodes (or electrode pairs) in a multichannel array;

5. Response fields found in a heterogeneous population of neurons are similar to the fields found for a uniform population for spatially-selective electrodes, but dissimilar in important ways for electrodes with large space constants;

6. Surprisingly, the "equivalent space constants" of the neural density profile found for monopolar stimulation can be much less than the space constants of the imposed electric field;

7. This may explain how patients implanted with monopolar arrays can rank their electrodes;

8. For patients with good nerve survival monopolar electrodes may preferentially excite dendritic nodes over the dynamic range of stimulation, while bipolar electrodes may excite dendritic nodes near threshold and axonal nodes thereafter;

9. It is likely that a higher bandwidth of temporal information can be represented to the central auditory system with dendritic activation than with axonal activation; and

10. Therefore a tradeoff between spatial selectivity and temporal bandwidth may exist for auditory prostheses.

In summary, initial application of the simplest of our ensemble models has been useful for demonstrating basic patterns of neural responses evoked by intracochlear electrical stimulation and for generating testable hypotheses of improved stimulus coding for cochlear implants. We will be evaluating the hypotheses implicit in points 3, 7 and 10 above in tests with future implant patients. In addition, we plan to develop and apply more-sophisticated ensemble models to elucidate details in the response fields not evident in the predictions of ensemble model 1. Results of these future studies will be reported as they become available.

E. Design of a portable speech processor

We have developed a portable, real-time speech processor appropriate for use with single-channel auditory prostheses. The main objectives of this effort were to (a) demonstrate that the fundamental frequency (F_0) of voiced speech sounds could be reliably extracted with a low-power processor for both noisy and quiet acoustic environments; (b) demonstrate that this processor could reliably mark and code the boundaries between voiced, unvoiced and silent intervals in running speech, in the same acoustic environments; (c) provide a "building block" for multichannel speech processors in which signals representing excitation of the vocal tract are coded separately from signals representing the "short-time" configuration of the vocal tract; (d) provide a working hardware system for implementing other promising strategies in a portable unit; and (e) make a prototype processor to provide speech input that is largely complementary to the input provided by information available in lipreading, primarily for application in extracochlear prostheses for infants and young children (after full evaluation of this and competing coding strategies with adults).

To meet these objectives we designed a portable processor based on the CMOS ("Complementary Metal Oxide Semiconductor," a low-power technology for integrated circuits) version of the INTEL 8031 microcontroller. This microprocessor has a 1 microsecond instruction cycle and on-chip peripherals that facilitate its use in a low-cost, battery-powered processor for real-time analysis of speech.

A block diagram of the current configuration of the hardware is shown in Fig. 1. The hardware consists of four main sections: the analog section for bringing speech from the environment to the input of an analog-to-digital converter (ADC); the ADC itself; the microcontroller section with memory; and a digital-to-analog converter (DAC) for output to the electrode driver(s). Under construction are two variations of this basic configuration, both to increase "processing throughout" with either the addition of another 8031 or a CMOS 12x12 bit multiplier (one of the ADSP-1000 series of multipliers made by Analog Devices, Inc.). These additional devices are not required for the present processing strategy, but may be required for more complex strategies such as those that might be used for multichannel prostheses. The power consumption of the present processor is about 70 mW for quiet environments, where not much current is drawn by the analog section, and about 74 mW when intense noise and speech are present at the microphone. At these power levels the processor will run continuously for several days on a 5-volt NiCad battery without recharging.

Also compatible with the objective of a portable unit is the small size of the instrument. Even with low-density construction, the entire processor easily fits on a 12x9 cm board. Improved packaging could easily reduce the size of the processor to that of a pack of cigarettes.

In addition to portability, another important objective of our effort was to extract a reliable and accurate representation of F_0 for voiced speech sounds. We selected the "Average Magnitude Difference Function" (AMDF) algorithm (Ross et al., 1974; Sung and Un, 1980; Un and Yang, 1977) because its computational complexity is relatively modest and its performance is robust in noisy acoustic environments (Paliwal, 1983). In

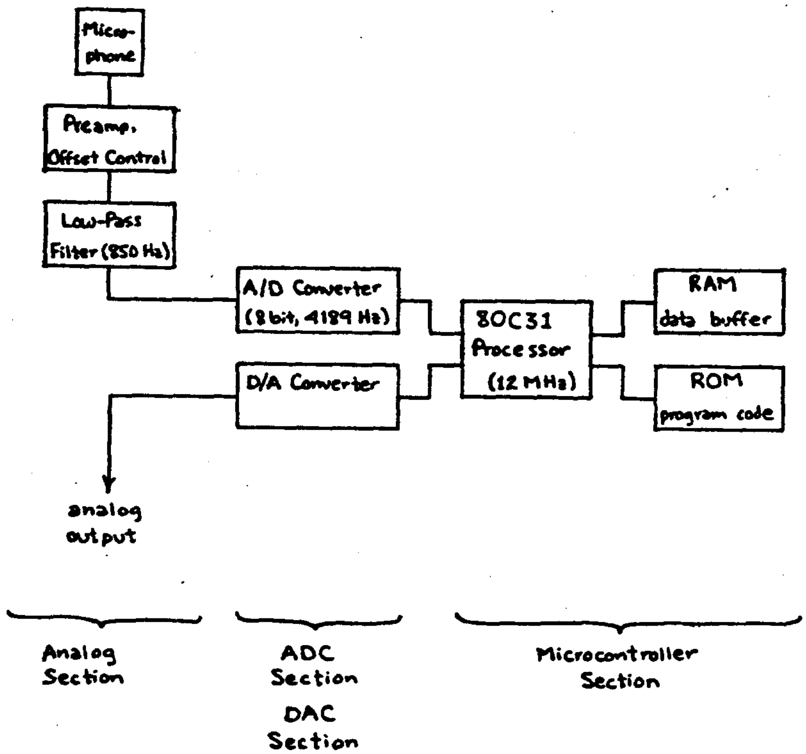

Fig. 1. Block diagram of the 80C31-based processor

our implementation of this algorithm the AMDF output is further processed for median smoothing, detection of erronous indications of F_0, and detection and signalling of intervals that contain unvoiced speech sounds. Tests with inputs of sinusoids, noise and speech material indicate that the processor functions according to its design. We therefore have available a portable processor for accurate extraction and representation of voice fundamental frequency.

F. <u>Evaluation of processing strategies in tests with patient LP</u>

In June and July of 1985 we conducted psychophysical and speech-testing studies with one of the percutaneous-cable patients at UCSF. Briefly, this patient (LP) presented a tremendous challenge to the UCSF and RTI teams in that his psychophysical performance along almost every measured dimension was worse than any previous patient in the UCSF experimental series. Moreover, only one of his scores on speech tests (voice/unvoice) was above chance for the "compressed analog outputs" strategy used in the present UCSF processor.

With these discouraging results in mind, our approach was first to reproduce one version of the analog-type UCSF processor in the software of our block-diagram compiler (see section II.B); then determine if the simulated processor produced results essentially identical to those obtained with the UCSF "tabletop" analog processor; and finally to evaluate other processing strategies in an attempt to improve LP's understanding of speech. The basic plan of these other processors was to reduce in steps the temporal and spatial overlap between channels and introduce (in the last two processors tested) a representation of the linear-prediction residual signal. In all, the block-diagram compiler was used to simulate 5 distinctly different processing strategies. As described in detail in our 7th Quarterly Progress Report, some of these processors produced percepts that were clearly in the "speech mode," that were spontaneously recognized as the speech test tokens delivered to the processor, and that produced test scores well above chance on confusion-matrix material. More specifically, the results of the tests with patient LP were consistent with the following:

1. Although some patients have good or excellent performance with multichannel processors that present "compressed analog" signals at their outputs (e.g., EHT, the previous experimental patient at UCSF), other patients have miserable performance with these processors;

2. The patients who have little or no recognition with the "compressed analog outputs" processor are likely to exhibit various manifestations of poor nerve survival and severe channel interactions;

3. In tests with patient LP, reduction of the spatial bandwidth of transmission of compressed analog outputs to the electrode array (by reducing the number of simultaneous output channels from 6 to 2) did not improve performance over the "6-channel, all outputs on" strategy (zero performance for both);

4. An immediate and compelling increase in speech recognition can be obtained (at least in LP and probably in patients with similar patterns of nerve survival) with a 6-channel strategy in which interleaved pulses are delivered to the two channels that have the highest RMS energy in each time frame (strategy 3);

5. Use of one representation of the linear-prediction residual signal (the "multipulse excitation sequence," as applied in strategies 4 and 5) can improve the "naturalness" of speech percepts and apparently also can convey much information over a single channel of stimulation;

6. However, performance on speech tests is in general lower with the "multichannel, multipulse excitation" strategies than with the "multichannel, interleaved pulses" strategy, possibly because the RMS energy levels in the selected channels are not represented in the former; and

7. It is likely, in view of points 1 and 4 above, that different processing strategies will be required for patients with different classes of neural pathology.

G. <u>Reporting</u>

A major effort of this project was in dissemination of research findings. In addition to our quarterly progress reports, we have prepared drafts of three manuscripts for publication, hosted a site visit for this project at RTI in December, 1984, and presented many papers at major conferences. The manuscripts in preparation are the following:

Wilson, BS and Finley, CC: Ensemble Models of Neural Discharge Patterns Evoked by Intracochlear Electrical Stimulation. I. Simple Model of Responses to Transient Stimuli. To be submitted for publication in Hearing Res.

Wilson, BS: Control of Temporal Channel Interactions in Multichannel Auditory Prostheses. To be submitted for publication in Hearing Res.

Weber, BA, Wilson, BS, Farmer, JC and Kenan, PD: The Duke University Cochlear Implant Program. To be submitted for publication in Am. J. Otol.

Finally, a list of major conference presentations is presented below:

Wilson, BS: Speech processors for auditory prostheses. Presented at the Neural Prosthesis Workshop, 1983, 1984 and 1985.

Wilson, BS: Coding strategies for multichannel auditory prostheses. Invited paper presented at the Gordon Research Conference on Implantable Auditory Prostheses, Aug. 19-23, 1985.

Finley, CC: An integrated field-neuron model of intracochlear stimulation. Invited paper presented at the Gordon Research Conference on Implantable Auditory Prostheses, Aug. 19-23, 1985.

Wilson, BS: Discussion Leader, Gordon Research Conference on Implantable Auditory Prostheses, Aug. 19-23, 1985.

Wilson, BS: Comparison of strategies for coding speech with multichannel auditory prostheses. Invited paper to be presented at the Conference on Speech Recognition with Cochlear Implants, New York University, April 17-19, 1986.

Finley, CC and BS Wilson: Models of neural stimulation for electrically evoked hearing. Invited paper presented in the special session on neurostimulation, ACEMB Meeting, Sept. 30-Oct. 2, 1985.

Wilson, BS and CC Finley: Speech processors for auditory prostheses. Invited paper presented in the special session on signal processing for the hearing impaired, <u>IEEE Bioengineering Conf.</u>, Sept. 27-30, 1985.

Wilson, BS and CC Finley: A computer-based simulator of speech processors for auditory prostheses. <u>ARO Abstracts</u>, 8th Midwinter Research Conference, p. 109, 1985.

Finley, CC and BS Wilson: An integrated field-neuron model of electrical stimulation by intracochlear scala-tympani electrodes. <u>ARO Abstracts</u>, 8th Midwinter Research Conference, p. 105, 1985.

Finley, CC: Co-chairman for session on Cochlear Prosthetic Devices, <u>ARO</u>, 8th Midwinter Research Conference, February, 1985.

Farmer, JC, Jr., Kenan, PD and Wilson, BS: Cochlear implants. Presented at <u>Surgical Grand Rounds</u>, Duke University Medical Center, November, 1985.

Wilson, BS and Finley, CC: Latency fields in electrically-evoked hearing. To be presented at the <u>9th ARO Meeting</u>, February, 1986.

Finley, CC and Wilson, BS: A simple finite-difference model of field patterns produced by bipolar electrodes of the UCSF array. Presented at the <u>8th IEEE-EMBS Meeting</u>, September, 1985.

III. Concluding Remarks and Recommendations

This has been a highly-productive project. In addition to meeting fully all requirements of the contract work statement with the completion of the activities described in sections II.A, II.B and II.E of this report, we have been able to (a) build powerful tools for understanding and defining the "electrical-to-neural transformer" that links the outputs of the speech processor to the inputs of the central nervous system (sections II.C and II.D); (b) compare the performance of competing processing strategies in tests with a single implant patient (section II.F); and (c) help establish new collaborative programs for testing implant patients at Duke University Medical Center and at Washington University Medical Center. Work in this first project with the Neural Prosthesis Program was primarily directed at development of tools for future work. These tools include the hardware interface (section II.A), the computer-based simulator of speech processors (section II.B), and the field-neuron and ensemble models mentioned above. Our goal for the next project is to apply the tools we have developed in the first project. We expect that patients in San Francisco, Durham and St. Louis will be included in an extensive series of tests, and that application of the above-named tools will allow us to improve significantly the performance of multichannel auditory prostheses.

IV. Acknowledgement

We are pleased to acknowledge the many contributions to this project made by our coworkers on the cochlear-implant team at UCSF. From the beginning they have offered assistance, encouragement and insights without hesitation. Moreover, we have been inspired by the quality and scope of their work on the development of auditory prostheses and by their determination to improve present devices. In retrospect, our progress on this project would have been far, far less without the enthusiastic support of the UCSF team.

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Appendix 1: Key Contents of Quarterly Progress Reports for NIH Project N01-NS-2356, "Speech Processors for Auditory Prostheses"

Quarterly	Contents
1	Description of initial, set-up trip to UCSF; Initial plans of a collaboration at Duke University Medical Center.
2	Model of field patterns produced by intracochlear electrodes; Collaboration between UCSF, Storz and Duke; First descriptions of (a) the hardware interface for communication between the Eclipse computer and patient electrodes, (b) the software for the block-diagram compiler, and (c) the rationale of intracochlear EP measurements.
3	Brief descriptions of hardware interface, computer-based simulator, DCU control software; Initial description of the Frankenhauser-Huxley model component of the Integrated field-neuron model.
4	Overview of first-year effort; Complete descriptions on (a) the development of the Frankenhauser-Huxley Axon component of an integrated field-neuron model of intracochlear electrical stimulation and (b) the use and application of a computer-based simulator of speech processors for multichannel auditory prostheses.
5	Description of spiral-plane calculations with the finite-difference field model; Initial examination of the effects of bone-fluid interfaces on current densities in the scala tympani and excitable tissue.

6 Development of portable, real-time hardware; Description of software for support of the RTI Patient Stimulator; Description of software for support of basic psychophysical studies and speech testing; Outline of Proposed tests with the present implant patient at UCSF, for the months of June and July, 1985.

7 Speech-testing studies with patient LP; Appendix on present status and functional description of the block-diagram compiler.

8 Development and application of ensemble models of the spatial and temporal patterns of neural discharge produced by intracochlear electrical stimulation.