



RESEARCH TRIANGLE INSTITUTE

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PART I

TECHNICAL PROPOSAL B

Speech Processors for Auditory Prostheses

Submitted in response to NIH RFP No. NIH-NINCDS-85-09.

POST OFFICE BOX 12194 RESEARCH TRIANGLE PARK, NORTH CAROLINA 27709

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*Denotes that this section and its subsections are identical in technical proposals A and B.

I. Introduction

The project outlined in NIH RFP No. NIH-NINCDS-85-09 is directed at the improved design of speech processors for auditory prostheses. Briefly, the project requires collaboration with a team or teams of investigators to evaluate processing strategies, and further requires the selected contractor(s) to (1) "design and develop a computer-based, multichannel waveform generator," which when coupled with the collaborating investigator's multichannel neural stimulator will permit basic psychophysical studies on improved methods of stimulus coding; (2) "design and develop a computer-based, multichannel auditory signal processor for use in evaluating promising speech extraction and stimulus encoding schemes;" (3) "design and fabricate wearable speech processors based on the results obtained with the computer-based simulated designs;" (4) "supply at least two of these wearable speech processors to the Project Officer by at least three years after the start of the contract;" and (5) "assist the collaborating human subject evaluation team in implementing the above mentioned waveform generator, computer-based signal processor and the wearable speech processors."

In this proposal we will first review our progress in the first 18 months of effort for NIH project N01-NS-2356, "Speech Processors for Auditory Prostheses." This project, of course, is the predecessor to the one described in the present RFP. In the current contract we at the Research Triangle Institute (RTI) have had the good fortune to establish a most productive collaboration with the excellent cochlear-implant group at the University of California at San Francisco (UCSF). The main activities of our combined team during the initial period of project effort were 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 evaluate 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, the product of which is a detailed set of plans for experiments to evaluate "stimulus primitives," single-channel coding strategies, and multichannel coding strategies;
4. Build a computer-based simulator that is capable of rapid and practical ~~evaluation~~^{emulation} of most of these approaches in software;
5. Conduct tests with an implant patient at UCSF to confirm proper operation of the equipment and software indicated in points 1 and 4 above, and to obtain measures of basic psychophysical performance with the UCSF transcutaneous transmission system;
6. Design and build a portable, real-time speech processor for single-channel auditory prostheses, based on use of the "AMDF" algorithm for extraction and presentation of voice pitch and voice/unvoice boundary information;
7. Help to establish a strong collaboration between UCSF, Duke University Medical Center (DUMC) and RTI, so that parallel series of tests with implant patients can be conducted in the immediate future at both UCSF and DUMC.

In all, we are proud of the progress we have made in the first 18 months of project effort. In addition to meeting fully all requirements of the contract work statement with the completion of tasks 1, 3, 4 and 6, we have been able to build a powerful tool for understanding and defining the "electrical-to-neural transformer" linking the outputs of the speech processor to the inputs of the central nervous system (task 2) and we have been able to help initiate a parallel testing effort at Duke (task 7).

Our primary goal for the work described in this proposal is to define the classes and parameters of processor design that will allow full recognition of speech without lipreading for recipients of multichannel implants in whom survival of peripheral dendrites is good. Other important objectives of the work we propose include (1) development of improved

strategies for implant recipients in whom survival of peripheral dendrites and/or ganglion cells is patchy or poor; (2) development of objective tests to determine the pattern of nerve survival in implanted patients; (3) further development of portable, real-time processors to implement particularly-promising strategies in "take home" units for implant patients; (4) investigation of learning effects (mainly by the collaborating psychophysical teams) with these take-home units; (5) development of improved strategies for coding speech information with single-channel, extracochlear auditory prostheses, primarily for the safe (and hopefully efficacious) use by infants and young children; and (6) development of improved understanding of the encoding of electrical and acoustic stimuli at the auditory nerve, through the studies we will describe on "stimulus primitives."

This is the second of two proposals we are submitting in response to this RFP. The first proposal outlines a project that will meet the requirements of the RFP at approximately our present level of effort, and the second proposal outlines an "expanded-scope" project in which the RTI team will (1) increase its effort in supporting the teams at UCSF and Duke, to evaluate processing strategies and stimulus primitives; (2) help initiate and continue to support a new evaluation effort at Washington University and Central Institute for the Deaf; (3) thoroughly evaluate more alternative designs for portable, real-time processors than the necessarily limited number described in the "present-scope" proposal; and (4) provide additional take-home processors for patient use and for further evaluation of possible learning effects.

To save time for readers of these proposals, we want to mention that both proposals have the same organization and that many sections are identical in the two documents. The identical sections are marked with asterisks on the CONTENTS pages of each proposal. In the background sections of both proposals we describe tools we have developed in the first period of our present contract. These include the hardware interface for communications between the Eclipse computer and implanted electrodes; the models of field patterns and neural discharges produced by stimulation with intracochlear electrodes; and the computer-based simulator of speech processors for auditory prostheses. In the background section we also present preliminary results obtained from the field-neuron model and from the tests with the implant patient at UCSF. Finally, we describe the

collaborations we have established with UCSF, DUMC and Storz Instrument Company, and we describe our work on development of portable, real-time hardware for a single-channel auditory prosthesis.

The next major section of both proposals is the "plan of proposed effort." Here, all subsections are the same with the exception of the one on further development of portable, real-time processors. The identical subsections describe (1) the design of stimulus primitives, single-channel coding strategies and multichannel coding strategies; (2) our suggested experimental plan to assist the collaborating psychophysical teams in evaluating these stimulus primitives and coding strategies; and (3) our view of the prospects for the proposed projects.

As might be expected, there are large differences in the sections on "Project Organization and Management" and "Statement of Work, Schedule and Budget." These sections present in detail the distinctions between the two projects.

Finally, the last sections, on protection of human subjects, literature references and RTI experience, are the same in both proposals.

II. Background

A. Patient Stimulator and Computing Hardware

The RTI Patient Stimulator is a flexible research tool, designed to allow great flexibility of stimulus control for speech processor experiments in cochlear implant research. The stimulator functions as an extension of the RTI Block-Diagram Compiler software, allowing full functional realization of the speech processors designed with the compiler. Its large dynamic range and bandwidth capabilities make the Patient Stimulator a transparent component in the simulation of a wide range of speech processors. Patient safety has been heavily emphasized in the design. An added feature is the ability to measure intracochlear potentials via nonstimulated electrodes within the implanted electrode array. This option has been included to allow more direct assessment of the correlations between applied electrical stimuli and elicited neural activity. A functional overview of the interface's capabilities, a review of patient safety features, and a discussion of the intracochlear potential measurement system follow below, along with a brief description of the computer hardware and software which control the patient interface.

1. Functional Overview

Designed as a transparent element in the testing of speech processor systems with patients, the RTI Patient Stimulator is a research tool for the laboratory environment. No attempt has been made to address the issues of the portability that will ultimately be required in useable patient devices. Rather, the stimulator design has focused upon maximizing the opportunity afforded by the percutaneous cable to deliver highly controlled stimuli, as well as to assess the physiological events underlying electrical stimulation within the cochlea. The stimulator also can simulate the poorer performance characteristics of hardware drivers used in current prosthesis designs, thus allowing full emulation of presently realizable systems.

Figure II-A-1 shows the three hardware units that comprise the complete

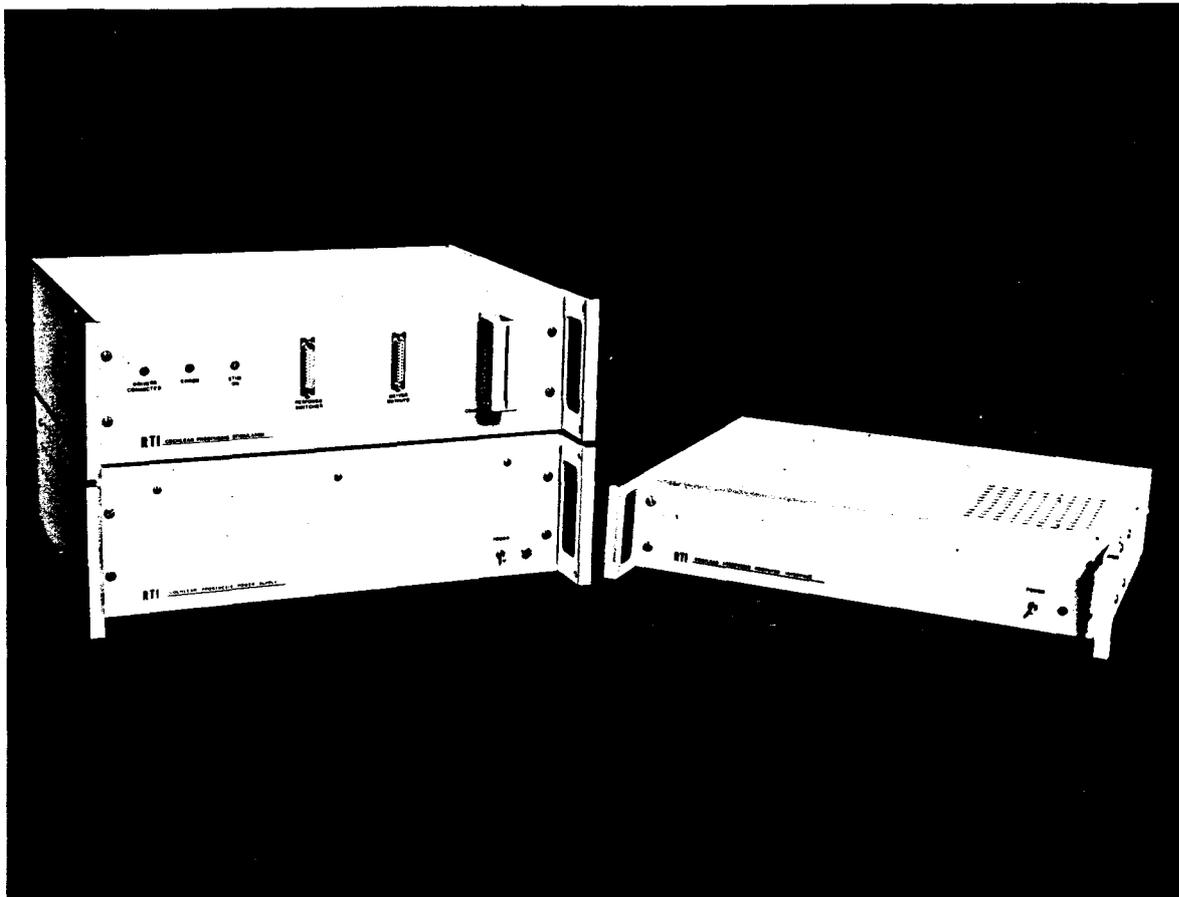


Fig. II.A.1. RTI Patient Stimulator System - On the left is the stimulator unit sitting atop its power supply unit. On the right is the logic interface unit which couples the stimulator system to its controlling computer.

stimulator system. On the left is the main stimulator unit, atop its associated power supply. These two units reside in the testing location with the patient. Three front panel connectors can be seen on the stimulator itself. On the left is the connector for a patient response panel. In the center is the connector for the cable leading to the patient's electrodes. Finally, on the right is a configuration jumper plug for establishing connections between stimulator channels and implanted electrodes. Three lights on the left end of the stimulator indicate various status conditions to the experimenter.

Up to 100 feet of parallel bi-directional logic line connect the stimulator with a logic interface unit, shown on the right in Figure II-A-1. This unit couples the stimulator's communications logic bus to the digital control unit (DCU) of a Data General Eclipse computer system. The logic interface unit is physically located with the Eclipse computer. The DCU is a separate programmable processor, capable of sharing memory with the Eclipse via data channel communications. It is connected both to the Eclipse data bus and to its own data bus. A standard parallel-bus interface card connects the DCU data bus to the logic interface unit of the prosthesis stimulator.

Figure II-A-2 is a block diagram of the computer facilities at UCSF, RTI, and DUMC that run our patient testing software. The lower portion of the diagram details the patient interface connections to each of the Eclipse computers. All three laboratory systems have been similarly configured to insure transportability of software. Horizontal dotted lines indicate communication links between the systems. Bulk data and software transfers are handled by magnetic tape, whereas daily maintenance, communication and coordination activities are handled by modem/telephone links. A more complete description of the computing facilities may be found in section IV.B., Facilities.

Figure II-A-3 is a functional block diagram of the stimulator system. The stimulator consists of six major functional subsections.

First are eight latched 12-bit multiplying digital-to-analog converters (DAC) which may be set independently under program control. Each DAC reference input may be connected to either an internal 10 V reference or an external signal source. DAC outputs are routed to the back panel of the stimulator where they are available as sources to external equipment

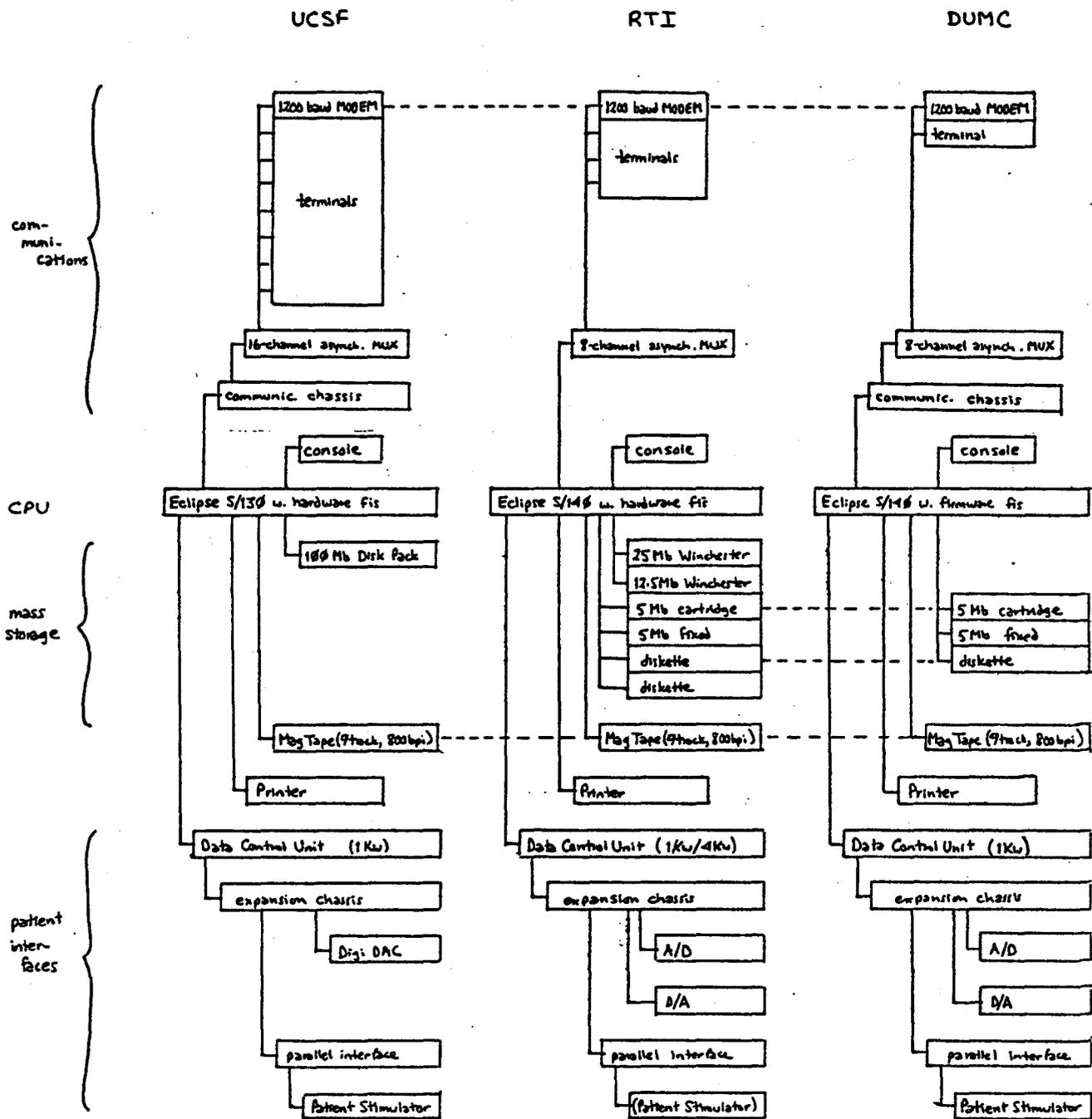


Figure II-A-2. Computing facilities.

(acoustic monitors, anti-aliasing filters, stimulus drivers for animal experiments) or are redirected as inputs to the stimulus drivers.

Second are eight high-voltage, wide-bandwidth op-amp driver stages that receive input from either the internal DAC's or external sources (anti-aliasing filters, etc.). Each driver stage is electrically isolated^{by} a wide-bandwidth (60KHz) isolation stage. Each driver may be operated in either a voltage-controlled mode or in a current-controlled mode. Output voltage compliance is + or - 70 volts. Peak output current levels are limited to a preset level (in the range of approximately 1 to 2 mA).

Third are separate, isolated power supplies (+ and - 70 volts) for each driver stage. Grounds for each stage are maintained separately.

Fourth are two methods of connecting the driver outputs to the patient electrodes. In both cases all driver outputs are routed through a series of patient connect/disconnect relays, allowing rapid interruption of all electrical connections to the patient. These relays may be operated under program control, manually by the patient, or automatically by the interface itself when valid data ~~has~~^{have} failed to arrive from the computer within a specified period (typically 50 usec.). Following the connect/disconnect relays are a jumper plug and a relay matrix for configuring driver connections to the electrodes. The jumper plug provides manual selection of connections of drivers (8 total) to electrodes (17, including ground) in various configurations (such as monopolar, bipolar, pseudobipolar). The relay matrix allows configuration of electrode connections for two driver stages under program control. The latter feature facilitates electrode impedance measurements and allows speed and flexibility in channel interaction studies.

Fifth is a two-channel analog-to-digital converter system for electrode potential measurement. This analog sampling system is used in impedance measurements and intracochlear evoked potential studies. Further discussion of this subsystem is in the section below on Intracochlear Potential Measurements.

Sixth is the control logic interfacing the stimulator subsystems (described briefly above) to the DG Eclipse computer. Communications with the Eclipse computer occur via two separate unidirectional parallel buses. One carries 16 bits, plus handshaking, to the interface from the Eclipse DCU; and, the other returns 16 bits plus a clock interrupt to the DCU. Both buses are provided with line drivers, allowing the stimulator to be located

up to 100 feet from the computer installation. Within the stimulator logic system is a programmable clock that synchronizes all data transfers to the stimulator system. Each data transfer^s is made in response to an interrupt generated by the stimulator clock. In the event that the stimulator fails to receive a valid data transfer from the computer in one interrupt clock period, the stimulator automatically disables itself, disconnects the patient, and signals the DCU that an error has occurred. Programming provisions for the stimulator are described in Appendix 1 for the interested reader.

Figure II.A.4. is a view of the main stimulator unit showing its physical construction. Six circuit boards are shown, each constructed using insulation displacement wiring techniques. Board interconnections are via ribbon cables connected at the ends of each board. The boards, as viewed from back to front, include (1) two logic circuit boards, (2) one eight-channel DAC board, (3) one eight-channel driver board with optical isolator^s stages (shown with only two channels populated), (4) one relay matrix and (5) one interconnect board with coupling capacitors to the patient electrodes. Not shown is the analog-to-digital converter board which inserts between the driver board and the relay matrix. The configuration jumper plug is on the right end of the front panel. The patient electrode cable connects to the front panel ~~connector~~^{connector} marked "DRIVER OUTPUTS."

A summary of the electrical and functional features of the Patient Stimulator follows:

- a total of eight stimulus channels, each consisting of a computer-controlled stimulus driver;
- each channel may function in either a voltage-controlled or current-controlled mode with + or - 70 volts output voltage compliance;
- each channel features a large-signal bandwidth of at least 60Hz to 60kHz;
- each channel functions independently, with^s electrically floating grounds and isolated supplies;
- each channel uses a latched digital-to-analog converter (DAC) allowing updating of stimulus channels only when stimulus magnitude changes are desired;

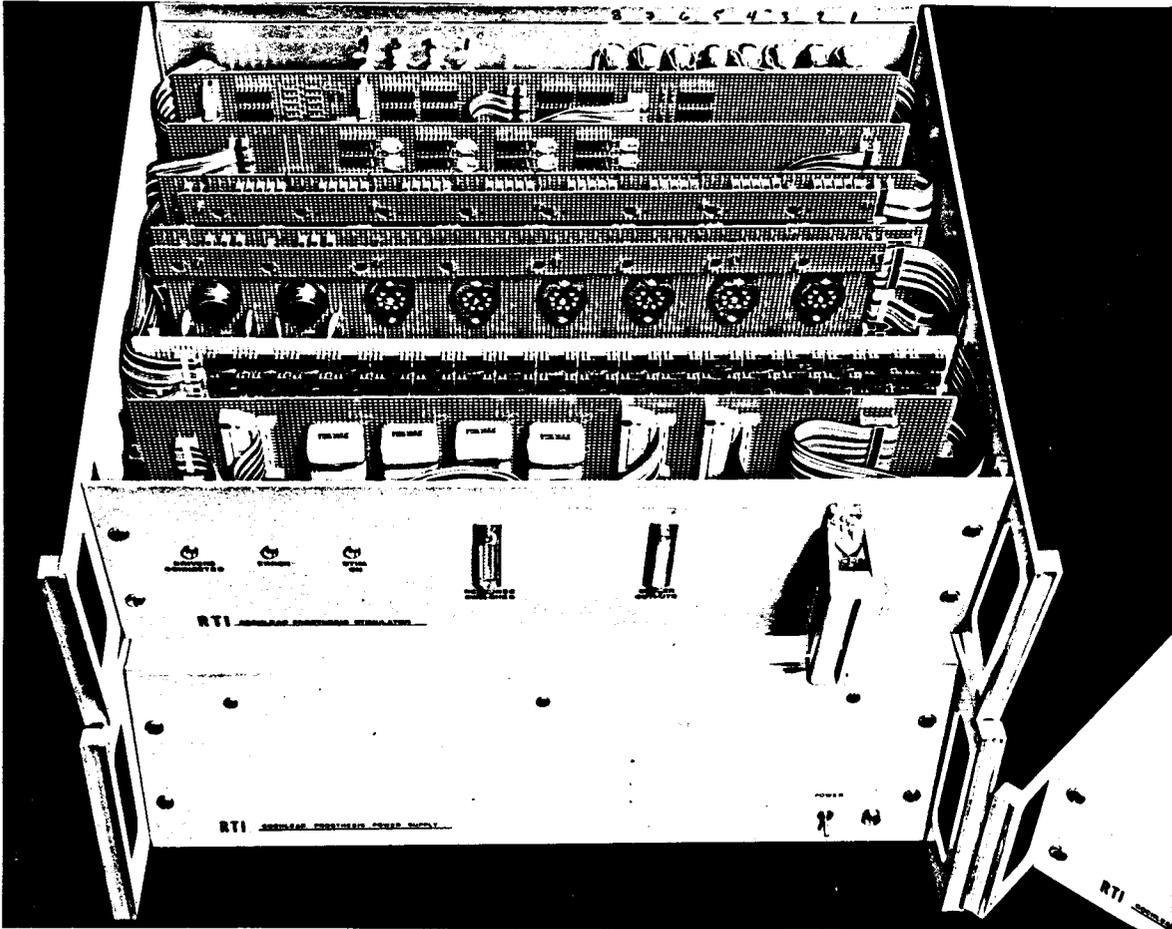


Fig. II.A.4. Patient stimulator with cover removed.

- all stimulus channel magnitude transitions are synchronized across all channels with a programmable timer built into the stimulator itself;
- electrode impedance can be measured between any two patient electrodes (when connected via a percutaneous cable directly to the electrodes);
- the patient and interface may be connected or disconnected under program control;
- connections between stimulus drivers and the electrodes may be achieved in two ways:

First, for complete flexibility under program control, two stimulus channels may be switched via a relay matrix to any combination of seventeen implanted electrodes, including a remote ground. This option facilitates characterization of all electrode impedances, as well as enabling rapid configuration of two stimulus channels for channel interaction studies.

Second, for rapid change of stimulator/electrode configurations, connections among up to eight stimulators and up to seventeen electrodes, including remote ground, are made by a front panel jumper plug. Each configuration plug is prewired for the desired stimulator/electrode configuration. Each configuration plug is also prewired with an identification number (0-32) which may be read and verified under program control during testing. This aids in preventing errors in changing stimulator/electrode configuration during evaluation of various speech processor designs.

- patient safety features are included (these are summarized in the next section);
- a two channel analog-to-digital converter (ADC) system is included for measurement of potentials appearing at the electrodes. This system allows both electrode impedance measurements and measurements of signals from unstimulated electrodes. The latter function is intended to be used in the measurement of intracochlear evoked potentials and stimulus artifacts. (A more detailed discussion of this measurement system follows in section

3 below.)

2. Patient Safety Design Features

Patient safety design features are listed here for review:

- optical isolation of analog circuitry;
- dual output blocking capacitors of low capacitance and low leakage;
- peak limiting of voltage and current stimuli are available at preset levels;
- patient disconnect relays on each channel operate under patient control or after a preset timeout period if communications with the controlling computer are interrupted or delayed;
- the high voltage supply for each patient channel consists of a standard modular supply, driven with an isolation transformer;
- the entire unit is further isolated with a medical-grade isolation transformer on the primary power support;
- standard maximum leakage current criteria for Subject Instrumentation Equipment are met or exceeded (lead leakage to ground <50uA).

3. Intracochlear Potential Measurements

A two-channel analog-to-digital conversion system has been included within the Patient Stimulator. This subsystem serves two functions. One is the measurement of electrode impedances. The other is the measurement of potentials from nonstimulated electrode pairs during active stimulation. This latter measurement technique will be used to assess the field patterns and evoked physiological responses occurring within the cochlea in response to stimulation. Below are brief descriptions of the hardware and software used for the impedance and intracochlear potential measurements. Techniques designed to eliminate the stimulus artifact component also are discussed.

Impedance measurements are conducted in a straightforward manner by injecting a small stimulus current and monitoring the consequent electrode voltage. One channel of the analog sampling system is connected by the programmable relay matrix across any two electrodes. Simultaneously, an

electrode channel driver is connected across the same two electrodes. Current is injected and the resultant potential is measured. Since all connections are under program control, measurement across many electrode combinations may be made rapidly. This process has been fully automated in software. By measuring the interelectrode impedances of at least three separate electrodes in all combinations, the impedance of each separate electrode may be calculated. Our impedance measurement software does this and presents characterizations of each individual electrode at various test frequencies and current injection levels.

Measurement of intracochlear potentials via nonstimulated electrode pairs is a more difficult task, but one that promises great rewards in understanding the physiological basis of intracochlear stimulation in an individual patient. Factors complicating these measurements include the small signal amplitude and the presence of stimulus artifact components. A variety of measures are available for use in circumventing these problems.

When making intracochlear potential measurements via nonstimulated electrodes a small op-amp head stage and switch matrix may be inserted between the patient's percutaneous cable and the cable from the patient stimulator. (Refer to the functional block diagram of the stimulator, Figure II.A.3.) The switch matrix allows the disconnection of the intracochlear electrode leads from the stimulator (thus significantly reducing capacitive loading and extraneous noise sources) and connects them directly to a differential op-amp buffer. The buffer head stage provides a high common-mode rejection ratio of at least 90 dB, input overvoltage protection up to + or - 36 volts and fast settling times (15 usec) after input overvoltages. Additionally, a programmable gain feature is included, providing preset gain settings (X1 - X1000) under program control.

Following the head stage is a sample and hold unit (S/H) that rapidly disconnects the signal input to subsequent filter stages, preventing latchup and slow recovery problems with high level stimuli. This S/H operates under program control and is typically used to disable the input signal path during high level stimulus periods. Immediately after the stimulus, the signal path is enabled and sampling is resumed. Subsequent to the S/H is a summing junction, allowing the injection of signals from an unused stimulus driver. This feature may be used as a cancellation tool in reducing unwanted artifact components during low to moderate level, continuous stimulation (for example, a sine wave stimulus).

A two pole bandpass filter (10 Hz to 5 KHz) follows for biological signal filtering. This filter also functions as an anti-aliasing filter in that the upper frequency cutoff of 5 KHz is a factor of four below the standard 20 KHz sampling rate of the ADC stage. All previous stages are powered by isolated supplies and all digital and analog signal lines are optically coupled.

A two channel 12-bit ADC completes the sampling system. All sampling and data transfer functions are controlled by the stimulator unit logic circuits. Sampling occurs synchronously with updating of stimulus driver channels, based upon the onboard programmable clock frequency.

Tests are presently being conducted with this unit to evaluate its performance in controlling artifact signals. Good performance is expected in that these techniques have been used with success in other recording situations. McGill, Cummins, Dorfman, Berlizot, Leutkemeyer, Nishimura and Widrow (1982) have recently reviewed the problem of neural recording with stimulus artifact suppression. In an experiment directly related to the intracochlear recording problem, Stypulkoski and van den Honert (1984) have recently succeeded in recording compound action potentials from monopolar electrodes in the cat modiolus during electrical stimulation.

B. Models of Electric Fields and Neural Discharge Patterns Produced by Intracochlear Electrical Stimulation

1. Overview

It is our view that the design of advanced speech processors for single and multichannel auditory prostheses is a two-part problem. First is the "classic" problem of extracting (or preserving) from speech those parameters that are essential for intelligibility. Second is the problem of transforming those parameters into electrical stimuli that will produce patterns of neural activity that are perceived as intelligible speech. Speech extraction is discussed in detail in Section III.B, Design of Single-Channel Coding Strategies, and in Section III.C, Design of Multichannel Coding Strategies. The present section describes our work on encoding the speech information into neural firing patterns.

Somewhat intermediate to the the extraction and encoding tasks described above is the determination of the neural firing patterns that best mimic those patterns elicited in a normal cochlea for a similar stimulus. We consider this task part of the design of encoding strategies. Further discussion on this topic is found in Sections III.B and III.C.

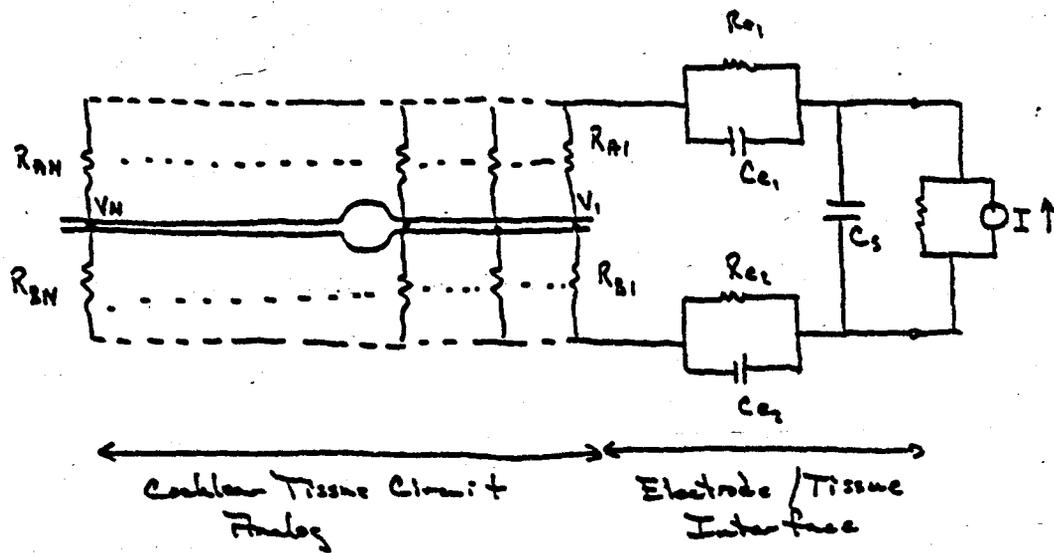
Consequently, for purposes of the present discussion, we define the encoding problem to be that set of issues that limit or influence the control that can be exerted over the firing patterns of VIIIth nerve fibers by electrical stimulation. In this context, we consider the encoding task to be a weak link in the design of any speech processor for a cochlear prosthesis and one that must be dealt with in an informed manner to ensure success in speech encoding. The total sequence of mechanisms that contribute to the electrical stimulation of neural elements within the cochlea is referred to as the "electrical-to-neural transformer".

Solution of the encoding problem requires detailed knowledge of the mechanisms of electrical stimulation of neural elements within the cochlea. Many factors are known to contribute. These factors include the physical locations, dimensions and electrical characteristics of the electrodes, as well as, the physiological integrity and survival patterns of the remaining neural elements. In addition, to achieve successful encoding of speech on

the VIIIth nerve, it is probably necessary to control both temporal and spatial profiles of neural discharge around each electrode or electrode pair. This task is further complicated by the presence of field interactions among the electrodes and the highly nonlinear response of the neural elements to electrical stimulation. To evaluate the relative significance of each of these contributing factors, we have developed a series of biologically-authentic computer models of the physical structures and biophysical mechanisms thought to be involved in the transduction of electrical stimulation to neural cochlear outflow.

Figure II.B.1 is a schematic diagram of the equivalent electrical circuit involved in the electrical stimulation of a single nerve fiber. This electrical analogue is the basis of our modeling efforts. More complicated arrangements which include multiple neurons and multiple electrodes are constructed upon this basic circuit. R_{e1} and R_{e2} are the series DC resistances of the electrodes 1 and 2, respectively. C_{e1} and C_{e2} are the electrode/tissue interface capacitances. Resistive dividers formed by the combination of R_{ax} and R_{bx} (x ranges from 1 to N) are distributed along the course of the neuron producing potentials (V_1 to V_N). The ratios of R_{ax} to R_{bx} , along with the electrode drive voltage, combine to produce the node voltages (V_N). C_s is the electrode cable shunt capacitance. The current driver is described as a Norton equivalent circuit whose parameters may be changed to represent a voltage-controlled, transcutaneous, radio-frequency-coupled link or the high-voltage compliance current sources used with the percutaneous cable. Observation of the model reveals that the absolute magnitude of the potentials along the course of the neuron is a direct function of the magnitude of the stimulus current passing between the electrodes; whereas, the relative ratios of the potentials along the neuron are a function of the physical geometry and resistive characteristics of the electrodes and surrounding tissue.

Estimates of the node voltages are calculated by an iterative, two-dimensional, finite-difference model of a cochlear cross section. The model contains a scala tympani bipolar electrode pair representing the UCSF bipolar electrode design, compressed into two dimensions (Loeb, Byers, Rebscher, Casey, Fong, Schindler, Gray and Merzenich, 1983). Resistivities are included in the finite difference calculations that approximate



Equivalent circuit of field calculation results.

Figure II.B.1

appropriate tissue characteristics. Fixed voltages are assigned to the electrode pair, and the outer boundary of the plane is held constant at zero potential level. Field patterns for the entire cross section are calculated by iteration. Those familiar with electromagnetics will realize that the computation as outlined is essentially an electrostatics problem. This formulation of the problem is appropriate since the tissue is predominately resistive and linear in nature up to 20 kHz. (Spelman, Clopton, and Pfingst, 1982). Finally, the potential levels at points along the locus of VIIIth nerve elements are extracted from the final field calculation. These potentials include the node voltages (VN) described above. A more complete description of this modeling approach is presented in section II.B.2. below.

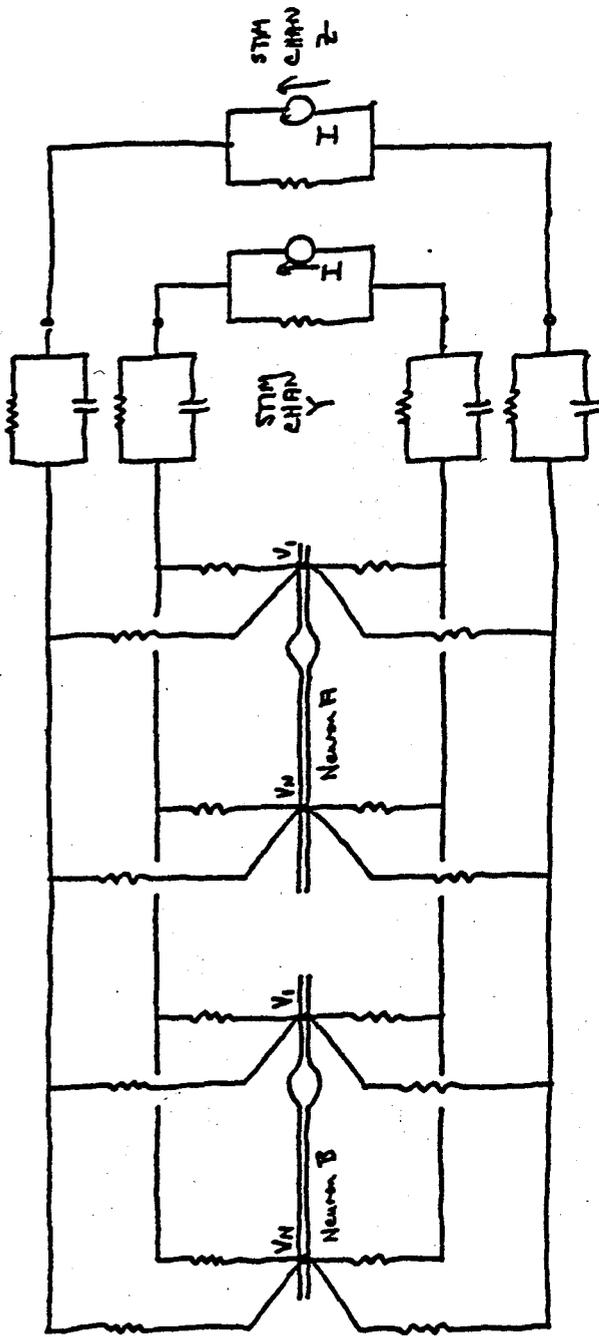
The motivating interest in the model is in predicting and studying the discharge characteristics of the neurons themselves. Consequently, the next step of modeling is the calculation of the neural responses to stimuli delivered by the implanted electrodes. This moves the modeling problem into the time domain and therefore links the modeling work directly to the issues of speech encoding strategies and the specification of stimulus current patterns. It is here that the modeling approach may prove especially valuable by providing insight into the factors controlling the temporal features of electrically-induced neural firing.

Temporal modeling is accomplished by feeding the calculated voltages along the neuron into a lumped-element model of a myelinated neuron. Stimulus inputs for the model are the potential profiles calculated in the field potential models described above. This model is a modification of McNeal's (1976) axon model consisting 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. The stimulus is 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.

The final step in modeling is to explore the interactions and

relationships between neurons both with single and multiple channel stimulation. Figure II.B.2 schematically outlines this approach by showing an integrated field-neuron model for a simple system composed of two neurons and two electrode pairs with current sources. Neurons A and B represent elements of an ensemble of neurons located at different positions within the spiral ganglion or along the basilar membrane. Each neuron has its own unique electrical relationship with each electrode pair as defined by the neuron's position in the field and the characteristics of the surrounding tissue. Furthermore, each neuron may have its own unique electrical characteristics, based upon its own biophysical features, such as diameter variation, loss or reduction of myelin, or even complete loss of peripheral segments. Stimulation with the current sources for channel Y and Z, either alone or simultaneously, allow study of the firing characteristics of both neurons A and B.

Full exploitation of these modeling tools will afford considerable insight into controlling the temporal and spatial profiles of neural discharge around an electrode or electrode pair. Additionally, insight may be gained into the complexity of field interactions occurring during multichannel stimulation, making it possible to avoid and/or exploit such interactions in speech processor design. Finally, appreciation of the deleterious effects of pathology may be factored into the speech processor design equation. This latter point may be the single most important contribution of the models, since interpatient variations in speech perception performance confound interpretation of test results as they apply to speech processor evaluations. Interpatient differences in pathology are prime candidates as sources of these variations. Clear insight into the details of electrical stimulation, specifically applied to intracochlear prostheses, can transform the stimulus encoding problem into a rational design task.



Generalized integrated field-neuron model
for two neurons and two electrode channel
stimulation.

Figure II.B.2

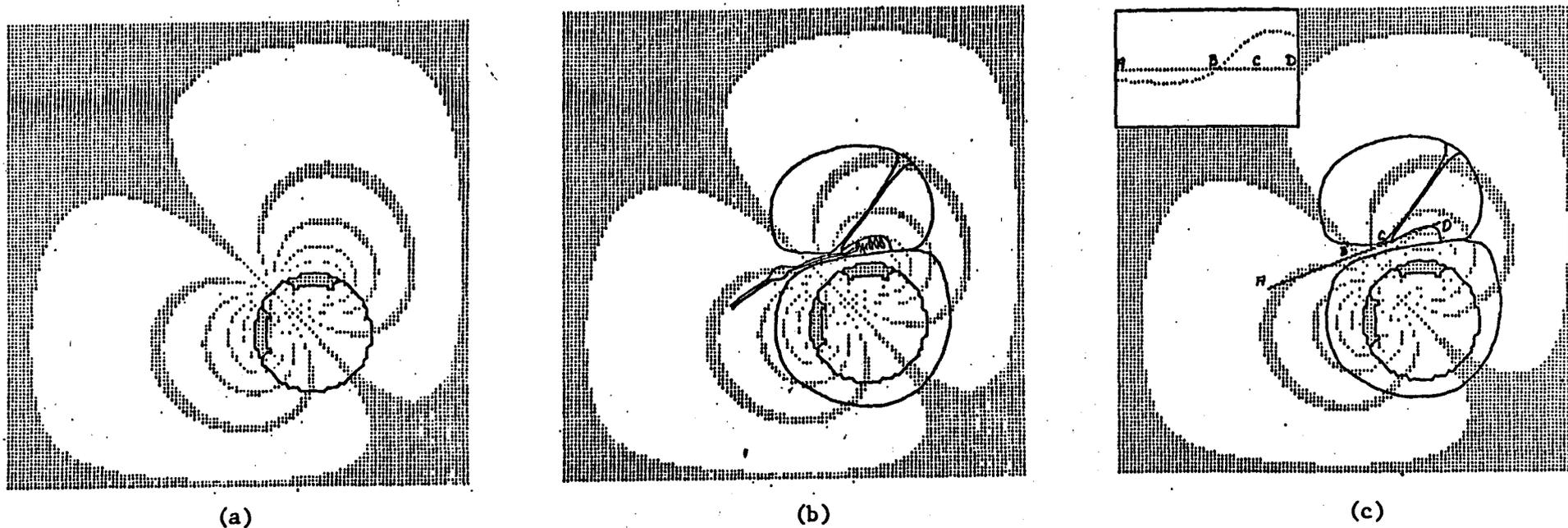
2. Models of Electrical Field Patterns within the Cochlea

The first model describes the electrical field patterns within the cochlea, as a consequence of stimulation by electrodes within scala tympani. The objective of this model is to provide estimates of the profiles of potentials along the loci of surviving neural elements. These potentials are calculated by an iterative, two-dimensional, finite difference model of a cochlear cross section, including a pair of electrodes in the scala tympani. Grid points in the model are 20 microns 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, the boundary of the grid is set to ground, and the resultant field patterns are computed by iteration for the entire cross section. Potential levels at points along the loci of the VIIIth nerve elements are extracted from the final field calculation. A detailed description of the two-dimensional cross-sectional model with initial results is given in Appendix 2. A portion of that presentation is included in the following discussion.

Figure II.B.3a shows the computed field pattern for a UCSF bipolar electrode pair compressed into two dimensions. Equipotential regions spaced at 10% increments of the full electrode potential difference (+ or - 1%) are plotted. These regions approximate equipotential contours in the tissue. Current flux patterns would be orthogonal to the equipotential contours.

Figure II.B.3b demonstrates the field calculation technique applied to a cochlear cross section. Cochlear tissue characteristics have not yet been included in this calculation. This figure thus shows the field potentials as they would be distributed across the cochlear tissues, assuming the tissue impedances have little or no effect. Once the model is fully completed, the cross-sections will replicate true histological cross sections and mimic the electrical nature of the various tissues. A finite-element modeling approach is being considered, in lieu of the present finite-difference method, in order to achieve greater resolution of tissue characteristics.

The purpose of the model is to determine the potential gradients along the course of the neuronal elements. It is the temporal and magnitude



Description of two-dimensional finite-element model.

(a) UCSF bipolar electrode pair and computed field pattern

(b) Electrode and field pattern with overlay of cochlear cross section

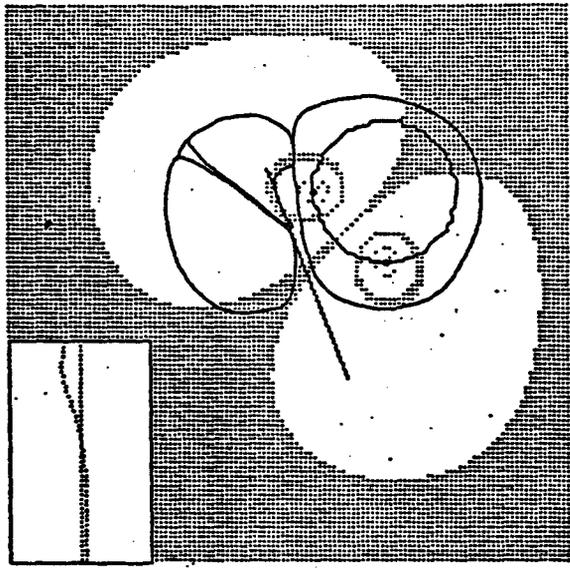
(c) Same as (b) but with inset showing potential levels along neuron (see text)

Figure II.B.3

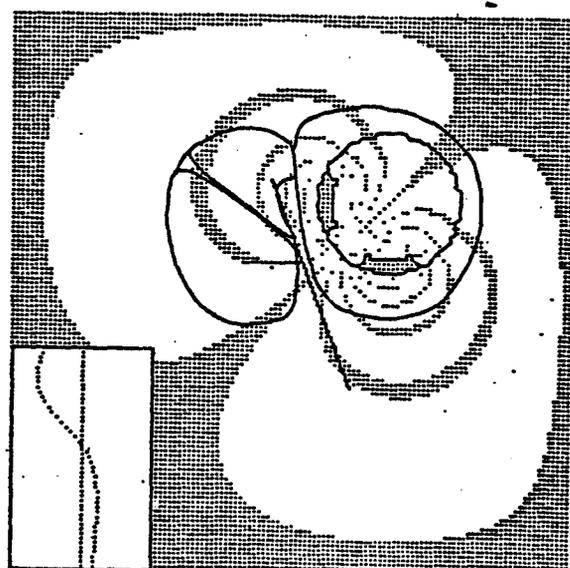
characteristics of these gradients that determine the behavior of the neural elements in response to the currents applied to the electrode pair. Figure II.B.3c repeats the presentation of Figure II.B.3b with the addition of an inset showing the calculated potentials along the locus of a neural element. The abscissa of the inset maps directly onto the straight line lying where the dendrites of VIIIth nerve neurons would lie. Points A-D indicate relative positions along this locus. The ordinate of the inset is voltage and ranges from the voltage (+V) applied to the upper (more lateral) electrode to the voltage (-V) applied to the lower (more medial) electrode. The horizontal line at mid-scale corresponds to a zero voltage level. Appendix 2 describes the effect of a variety of geometric manipulations of the electrode configuration and the resultant potential gradients along the neural elements. This ability to manipulate the physical features of the electrodes and evaluate the resultant effects on gradients in surrounding tissues affords a powerful tool for optimization that could otherwise only be achieved through lengthy and expensive experimentation with various implants and numerous patients.

Figure II.B.4 shows the field patterns for an actual bipolar pair and for a point source dipole model of the electrodes. At the locus of the neurons, the field profiles are different with the more realistic electrode model presenting larger potential gradients. This suggests that the field for a scala tympani electrode pair, placed close to the neural elements, may be poorly described by analytic models that assume dipole sources (Spelman, Clopton and Duckert, 1984; Soma, Spelman and Rubinstein, 1984). Rather, it appears that the neural elements actually lie in the near field of a well placed electrode. Consequently, dimensions such as electrode size and interelectrode distances are significant factors to be considered in the field analysis.

To illustrate the parametric power of the model, Figure II.B.5 shows a series of four calculations in which electrode placement within scala tympani has been varied. Figure II.B.5c shows the field patterns for an electrode optimally placed beneath the bony spiral lamina. In contrast, Figure II.B.5d shows the field condition with a poorly placed electrode pair that lies approximately 800 microns away from the bony spiral lamina. Note the substantial differences in the potential profiles along the neuron locus for the two conditions. Comparing the actual potential levels at node



(a)

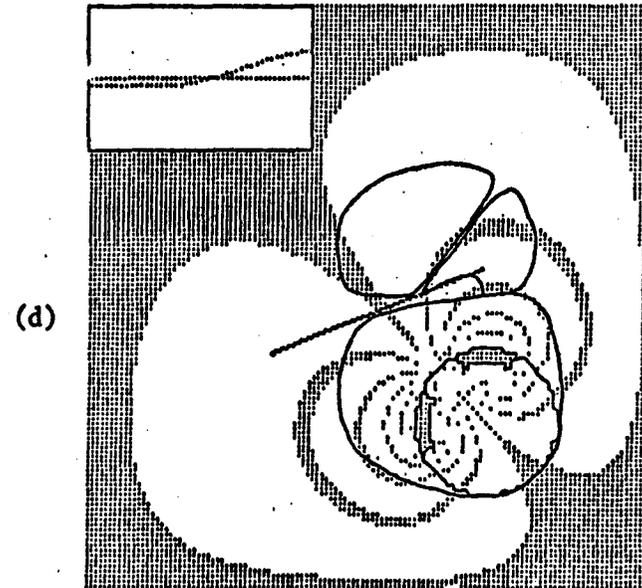
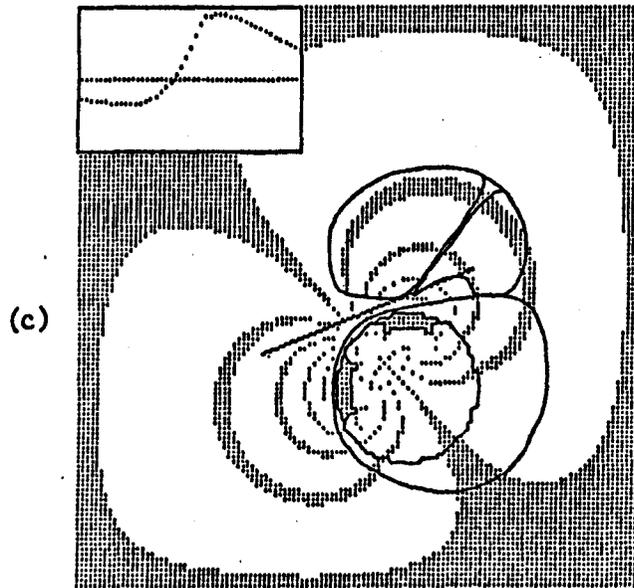
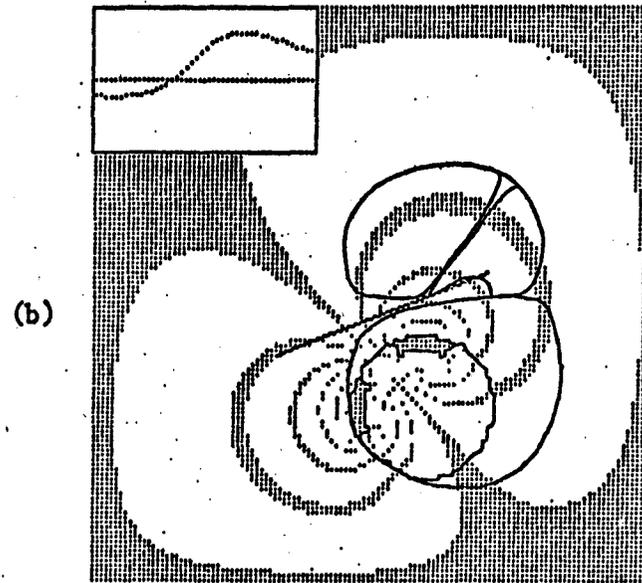
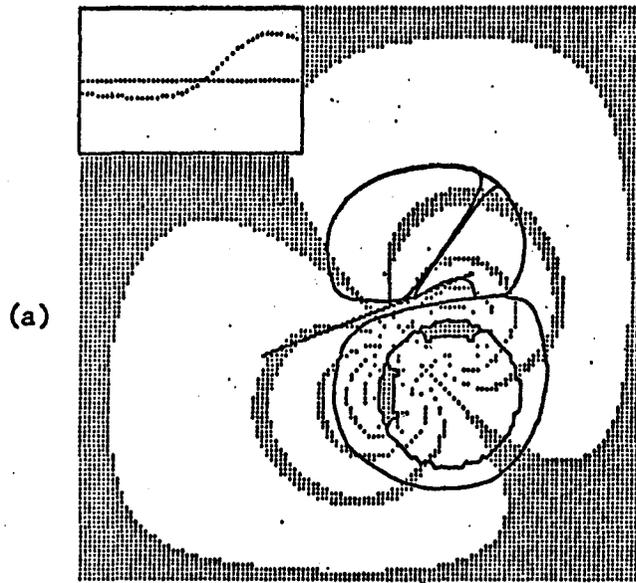


(b)

Field patterns for an actual bipolar pair (a) and a true dipole (b) configuration.

Figure II.B.4

Figure II.B.5



Field patterns for the standard bipolar pair located at different positions within scala tympani.

locations for the two conditions indicates that the stimulus generated by the poorly placed electrodes is about 12 dB below that generated by the well placed electrode pair. The model, as shown, is dimensionally similar to the condition of the cat cochlea. Calculated results agree well with measured data from the cat. Merzenich and collaborators found a 16 dB decrement in brainstem evoked potential thresholds with similarly placed electrodes in the cat (Merzenich, Leake-Jones, Vivion, White and Silverman, 1978).

It is also significant to note that a 16 dB decrement in stimulus sensitivity is effectively very large when considered against the 8 to 20 dB dynamic range for electrically evoked hearing. This suggests that particular attention should be given to controlling the position of implanted electrodes within scala tympani. Of the multichannel electrode arrays presently in use, the UCSF array, with its mechanical memory, would appear to be far superior to the thin, flaccid electrode arrays used by the Innsbruck, Melbourne and Stanford groups. These normally straight arrays assume the shape of the cochlear spiral only upon implantation where they come to rest against the outer wall of the scala tympani (Shephard, Clark, Pyman and Webb, 1985).

So far only field patterns for a bipolar electrode pair in the cochlear cross section have been considered. More global questions are of interest in the case of multielectrode arrays. In particular, the question of field overlap among stimulus channels is crucial to speech processor design, since discrete stimulation of separate sectors of the cochlear partition is necessary to access the tonotopic organization of the auditory system. A variety of electrode configurations (monopolar, bipolar, or pseudobipolar) are used by laboratories around the world. Debate continues on the relative merits of one method over the others. To guide our thinking on these matters, a second version of the field model has been developed. This model, containing the spiral of the cochlea compressed into two dimensions, is used to estimate the interaction and crosstalk at a single neural element for stimulation of two or more electrode channels. Calculations in this plane provide an estimate of field patterns over the entire extent of the electrode array, as opposed to the local field patterns of the cross-sectional calculations described above and in Appendix 2.

Figure II.B.6 shows the position of each of the sixteen electrodes of the UCSF electrode array when viewed from above the plane of the electrode

UCSF ELECTRODE ARRAY

showing spiral of mechanical memory,
positions of 16 electrodes, and
locations of 20 radial dendrites spaced at
1 mm. intervals along the spiral.

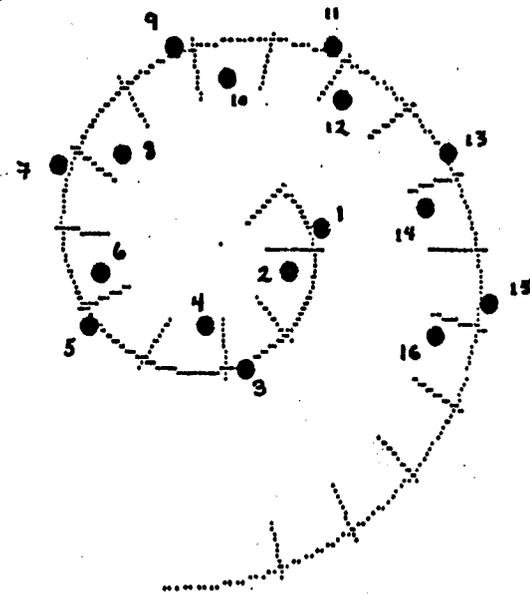


FIGURE II.B.6

array spiral. Positions are determined both by the electrode placement in the silastic carrier and by the spiral configuration of the array itself, the latter of which is determined by the assembly's mechanical memory. Each electrode is indicated by a closed circle, and the center line of the silastic carrier is shown by the spiral curve. Perpendicular to the spiral are short lines indicating the positions of radially-directed dendrites, spaced at one millimeter intervals along the spiral. Potential field patterns are calculated using the finite-difference method described above. The tissue medium for calculations in the plane of the spiral is assumed to be homogeneous and therefore the results do not reflect possible effects of impedance differences at tissue boundaries. Evaluation of this assumption of uniform tissue properties is addressed later.

Figure II.B.7 illustrates the results of a field calculation with the two most-apical electrodes energized. Electrode [1] is polarized with a positive voltage and electrode [2] is polarized with a negative voltage of equal magnitude. Computations are conducted with the outer boundary held to zero. The right panel shows isopotential contours in a subsection of the plane. Also shown are the electrode locations and spiral path of the electrode array. The locations of radial dendrites are labeled A-T, with an arrowhead pointing to the modiolar (or most medial) end of each dendrite. The left side of Figure II.B.7 indicates the potential levels along the locus of each dendrite, A-T. The ordinate of each small panel is voltage, ranging from the positive voltage magnitude at the most apical electrode [1] to the negative voltage magnitude at the next adjacent electrode [2]. The abscissa indicate positions along the dendrites.

Evaluation of the resultant potential profiles suggests that the fields in the vicinity of the apical stimulating pair are highly asymmetrical and also could stimulate a portion of the adjacent, more basal turn of the cochlea. To illustrate, dendrites D, E, L and M lie close to the 50% isopotential contour. Panels D, E, L and M show essentially zero stimulus voltage along the length of these dendrites. Therefore, these neural elements would be little affected by stimuli delivered to electrodes [1] and [2]. In contrast, dendrites H and Q are located at opposite poles of the bipolar stimulus field and consequently have significant and nearly-constant voltages imposed along their lengths. Responses of neurons at these locations could produce "turn-to-turn crosstalk" in percepts elicited by

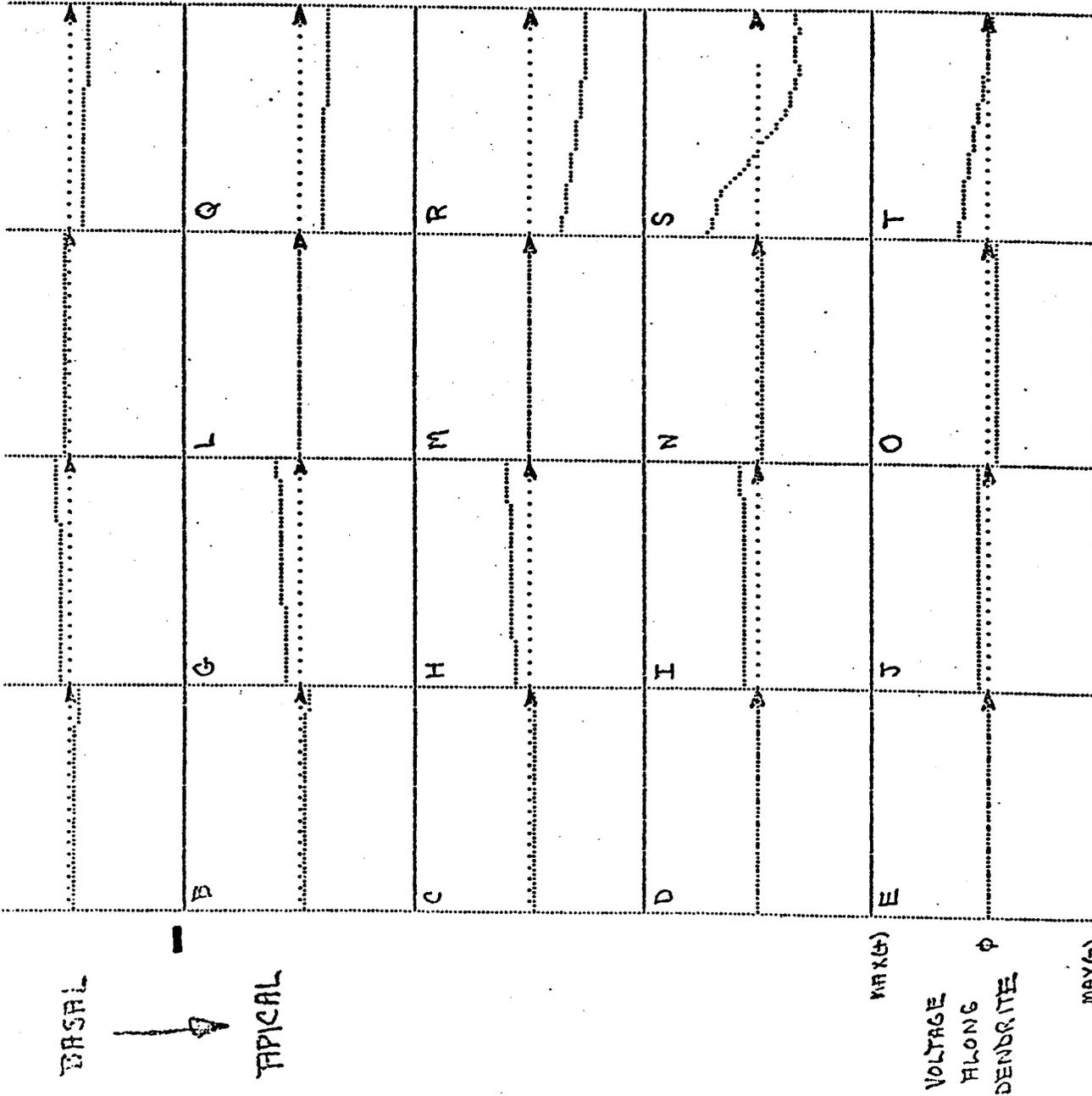
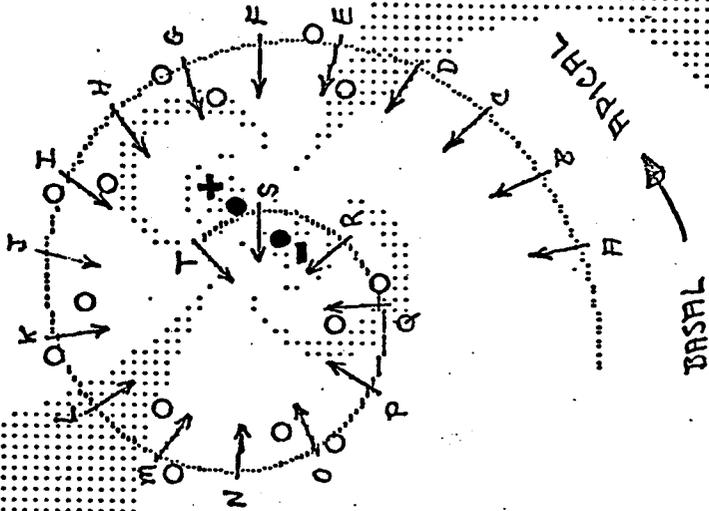


FIGURE II.B.7

relatively-intense stimuli. Finally, as expected, the greatest potentials appear in the immediate vicinity of the electrodes. Dendrite S, which is located midway between the offset bipolar pair, has a steep gradient from positive to negative along its length. Dendrites R and T, which are equidistant from the electrode pair along the basilar membrane, have less steep gradients and only one polarity of imposed potentials. If the magnitude of the imposed potentials along the dendrite is the excitatory aspect of the stimulus (as opposed to the voltage gradient along the dendrite or some combination of gradient and magnitude), as has been suggested by our applications of the myelinated axon model (see section II.B.3 below) and by the work of others with models of mammalian myelinated nerve (see, e.g., Ranck, 1975), dendrite R will have a threshold of response that is 2 or 3 times lower than the threshold of response for dendrite T. This difference of stimulus magnitudes is notable for these equidistant positions, indicating a significant asymmetry in the effective field of stimulation for the bipolar pair. This asymmetry is largely due to the curling of the cochlear spiral toward one of the poles of the bipolar electrode pair.

It should be noted that the above situation would most likely correspond to that of a patient in whom dendrite survival is good. The continued medial course of the neurons through the cochlea and into the modiolus is not depicted here. Modeling of this more complex situation must await expansion of the model to three dimensions.

Figure II.B.8 provides a high-resolution picture of voltage profiles produced along dendrites in the implanted ear for the stimulus conditions of Figure II.B.7. In Figure II.B.8, a continuum of dendrite positions is presented from the most-basal to the most-apical positions of the electrode array. The three panels show the extracellular stimulus voltage at the most-medial, the mid, and the most-lateral positions of the dendrites. The abscissae indicate positions along the basilar membrane from basal to apical. The midpoint positions of each bipolar pair are indicated by the vertical lines, with the relative position of each electrode indicated by its number. Midpoint positions of bipolar electrode pairs are 2 mm. apart along the basilar partition.

Figure II.B.9 shows the same potential distribution information as Figure II.B.8, except the potential distributions are shown for the

OFFSET BIPOLAR
CONFIGURATION
(1-2)

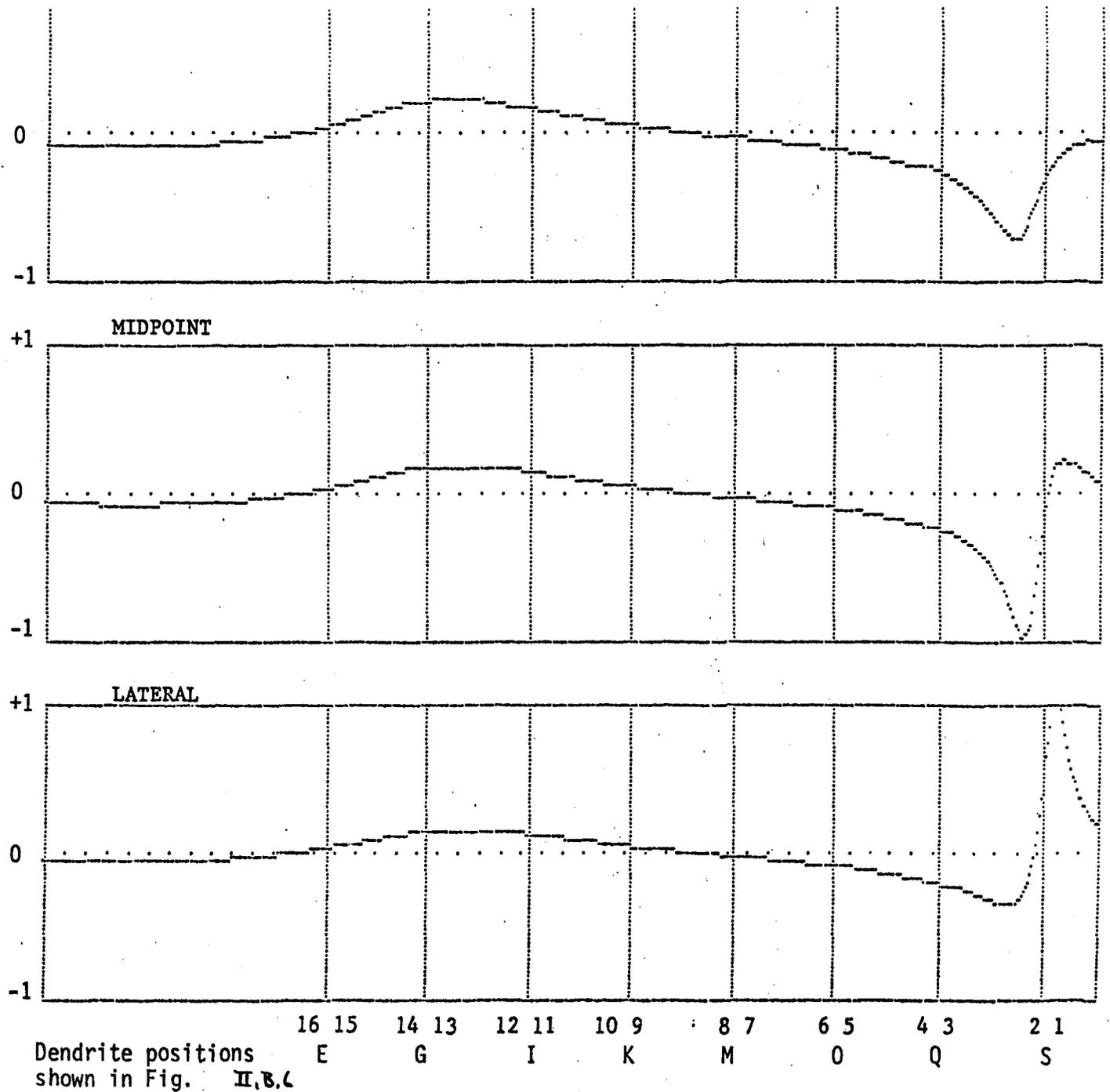
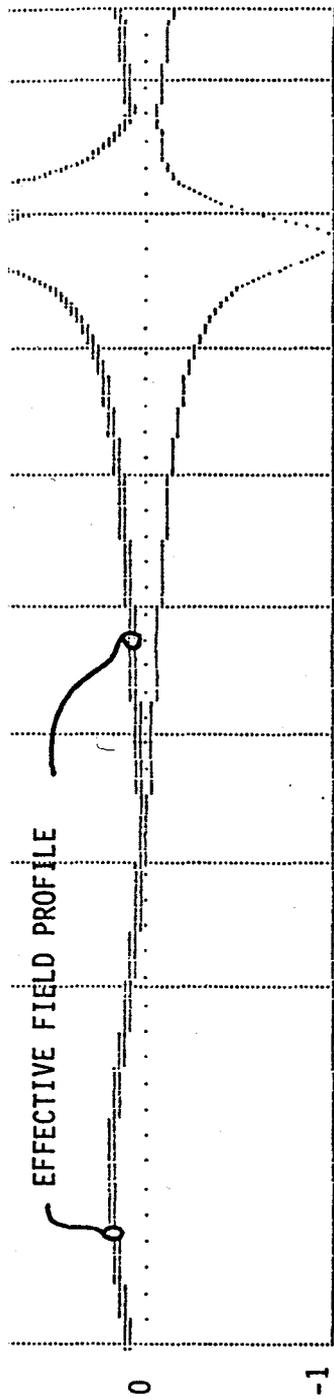


FIGURE II.8.8

condition of electrode [3] being positive and electrode [4] being negative. The distributions shown in Figures II.B.8 and II.B.9 are essentially equivalent with minor amplitude differences appearing in the medial and midpoint dendrite positions. These differences arise because of the more medial placement of electrode [4] relative to the spiral centerline than for electrode [2]. This is due to the tapering of the silastic carrier at the electrode array tip.

An additional detail seen in Figure II.B.9 is the double line shown in the upper portion of the top panel. This line depicts an approximation of the "effective stimulus field" of the electrode pair, driven with a balanced biphasic pulse. The effective stimulus field is best interpreted as a profile of the probabilities of firing for the ensemble of neurons affected by the stimulus. This profile is derived by plotting the peak, absolute values of the potential levels at the medial, midpoint and lateral dendrite positions, which roughly correspond to node locations along the dendrites. Absolute values are used to account for the effects of both phases of the balanced biphasic pulse. Peak values are used since the most strongly driven node along the dendrite will have the highest probability of firing. The representation thus obtained is a first approximation of the firing probabilities of neurons along the basilar partition in response to a balanced biphasic stimulus applied across a bipolar electrode pair. Temporal features of the stimulus and the neural response are neglected, and consideration of anatomical variations (e.g., fiber diameters) is not made. Tissue impedance effects have also not been rigorously modeled (see later discussion). Despite these inherent limitations of the present model, it is instructive to examine the derived effective stimulus fields of both single- and multiple-channel stimulation.

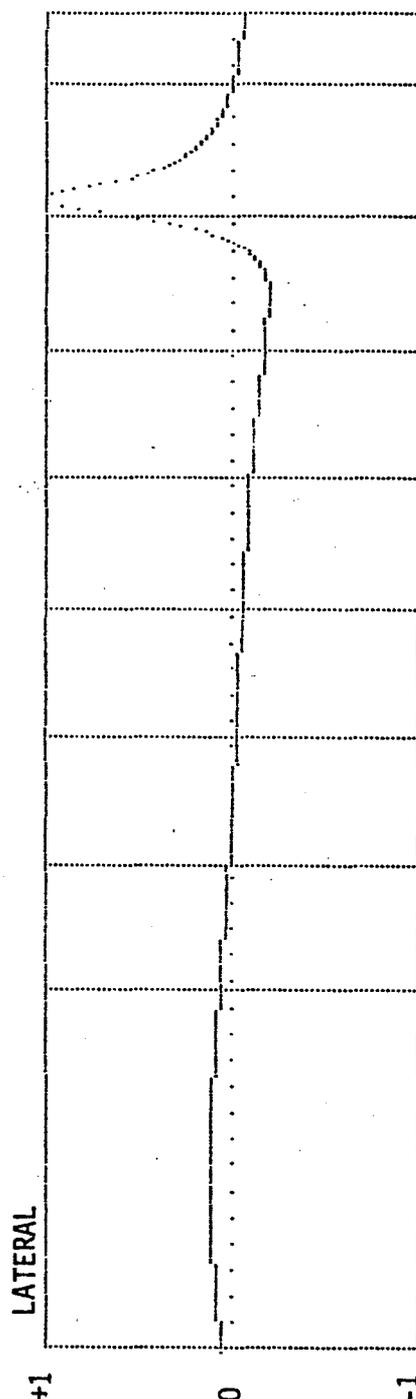
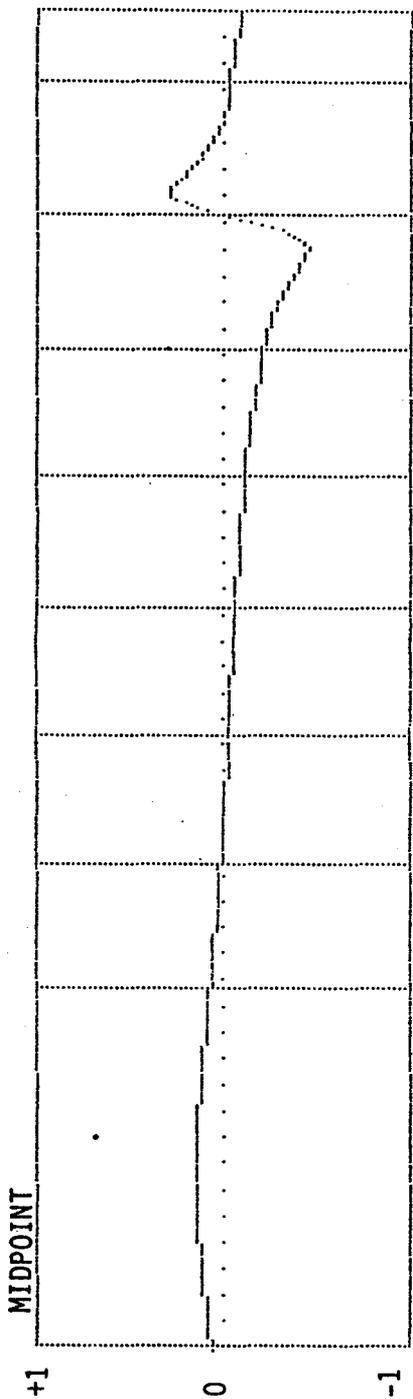
The effective stimulus field shown in Figure II.B.9 has a complex, double peaked profile with asymmetrical roll-off rates beyond the peaks. The double peaking is due to the offset configuration of each of the dipole pairs of the UCSF array. This follows intuitively if one considers that each electrode will have its own sphere of maximal influence in the region nearest the electrode and that the electrodes are physically displaced along the basilar partition. The asymmetrical roll-offs are largely due to the spiral shape of the electrode array which curls toward or away from the axis of the stimulating electrode pair.



OFFSET BIPOLAR

CONFIGURATION

(3-4)



6 5 4 3 2 1

FIGURE II, 8, 9

An interesting point that arises here is that substantial sharpening (narrowing) of the effective stimulus field of an offset dipole pair might be obtained by using an asymmetrical biphasic pulse, as compared to the effective stimulus field that occurs with a symmetrical biphasic pulse (i.e., one of the two peaks of the effective stimulus field could be "selected" with the use of asymmetrical biphasic pulses). A possible test of this hypothesis is to determine if a patient with good dendrite survival can distinguish between an asymmetrical, balanced biphasic stimulus and a similar stimulus, but of opposite polarity, applied to the same electrode pair. In such a patient, different populations of fibers (with some overlap) would be excited with each stimulus, and the resulting percepts should be distinguishable if our model of stimulus profiles is essentially correct.

Similar examination is instructive for the case of simultaneous stimulation of adjacent dipole pairs. This stimulus condition often results in considerable channel interactions in which stimulus fields for the adjacent dipoles appear to overlap. For the present discussion, only the situation of simultaneous stimulation with symmetrical biphasic pulses is considered. Figure II.B.10 shows the potential profiles when both dipole pairs are stimulated simultaneously. This is equivalent to summation of the data shown previously in Figures II.B.8 and II.B.9. Note that the ordinate range in Figure II.B.10 is twice the range shown in Figures II.B.8 and II.B.9. The effective stimulus field for the combined stimulation is also shown in Figure II.B.10 as the double line. Figure II.B.11 shows the opposite situation in which the stimulus polarity of the biphasic pulse applied to electrodes [1] and [2] has been reversed. Again, the effective stimulus field is indicated by the double line. One point to note is that the effective stimulus fields in Figure II.B.10 and II.B.11 are identical. This result is a simple consequence of superposition of the stimuli, and is consistent with the results of channel-interaction experiments conducted with patients in whom dendrite survival appeared (by several measures) to be good. That is, for the good-survival case one would expect the threshold of responses to in-phase stimuli delivered to the two electrode pairs to closely approximate the threshold of responses to out-of-phase stimuli. This expectation is, in fact, borne out in the experimental results for such patients. However, for the case of poor or patchy survival of dendrites,

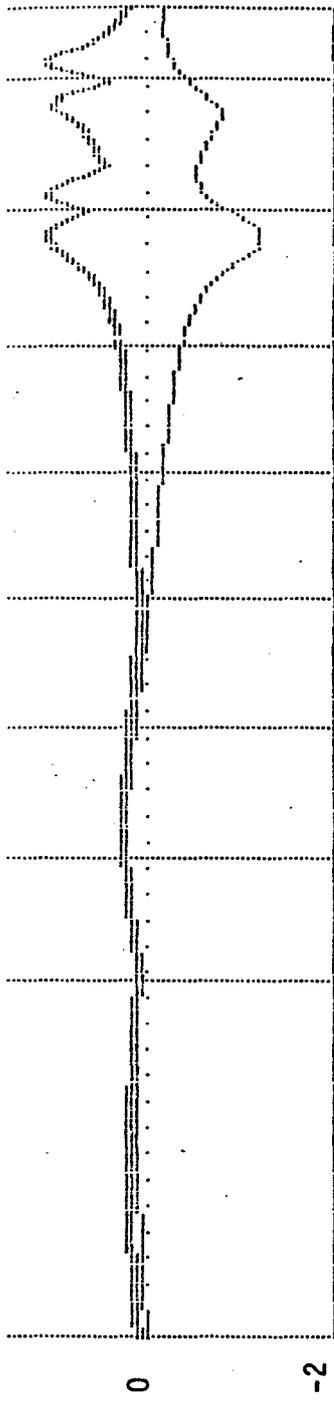
OFFSET BIPOLAR

CONFIGURATION

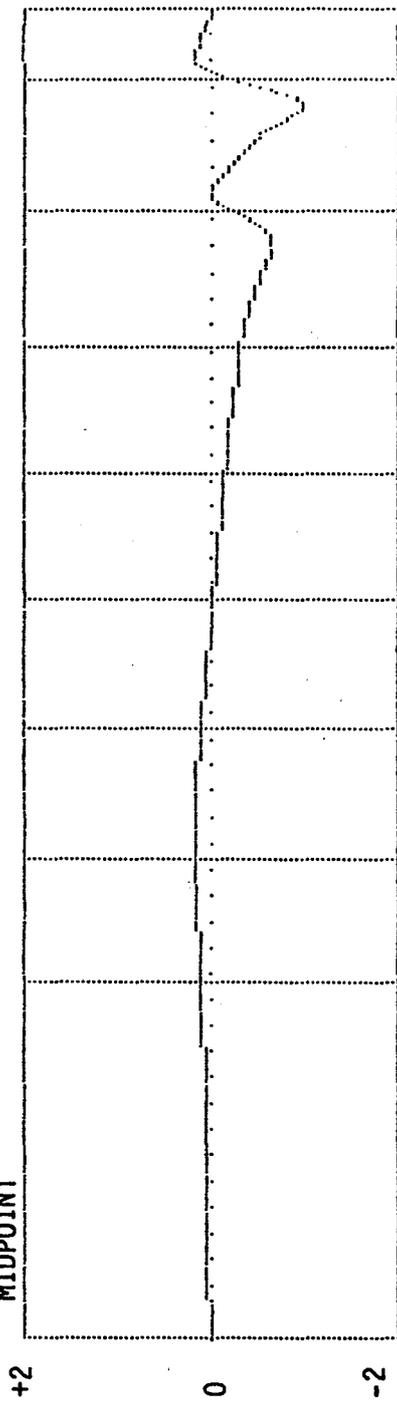
(3-4)

plus

(1-2)



MIDPOINT



LATERAL

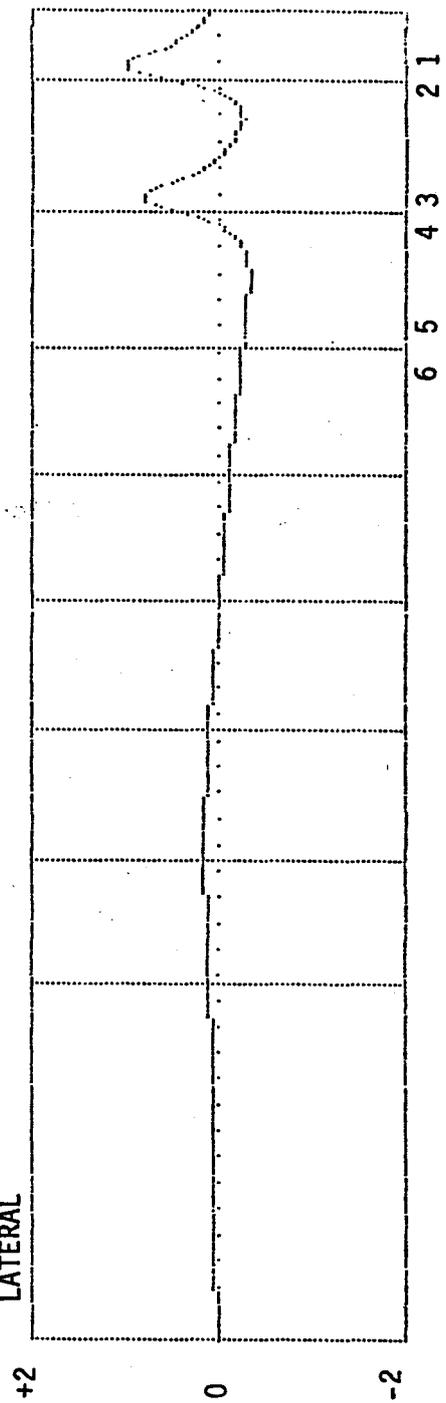


FIGURE II, B, 10

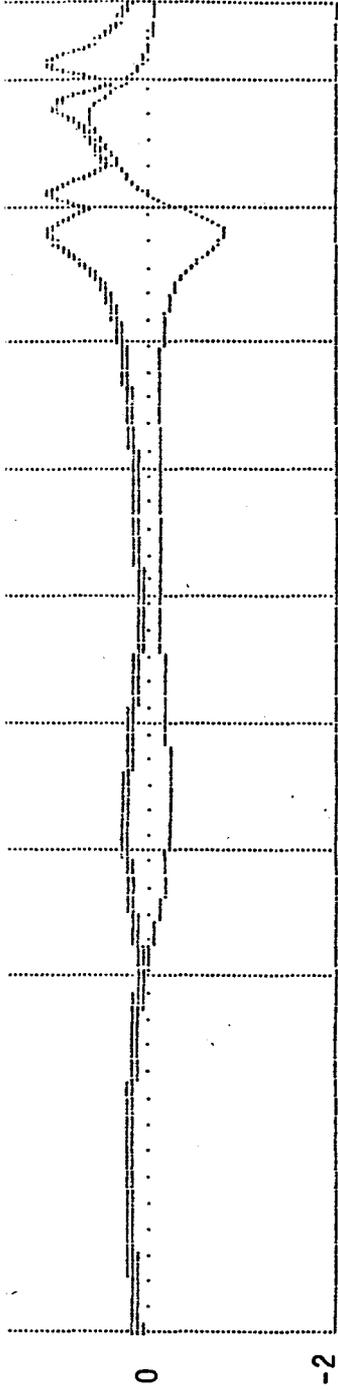
OFFSET BIPOLAR

CONFIGURATION

(3-4)

minus

(1-2)



MIDPOINT

+2

0

-2

LATERAL

+2

0

-2

6 5 4 3 2 1

FIGURE II. B. 11

the effective stimulus field will no longer correspond to that plotted in Figures II.B.10 and II.B.11 because the absence of stimuable neural elements at certain locations will skew the population of responding neurons. This skew would manifest itself in the form of significant channel interactions and significant differences in the thresholds to in-phase and out-of-phase stimuli. These predictions are entirely consistent with the results of channel-interaction experiments conducted with patients in whom dendrite survival appeared to be poor. Therefore, channel interactions and related phenomena are the likely result of uneven survival of dendrites, in contrast to current explanations in terms of significant overlap of stimulation fields.

Although the results presented above have clear significance for the design of speech processors for auditory prostheses, it is important to remember that several assumptions underlie the model in its present, simplified form. The major assumption of the spiral-plane model is that the characteristics of tissue in the plane of computation are homogeneous. An obvious method for evaluating this and other assumptions in the model is to compare model predictions with the results of animal experiments in which direct measurements of stimulus-response fields can be made. One such set of experiments was performed at UCSF in the late 70's with an array of aligned bipolar electrodes placed in the scala tympani of adult cats (see, e.g., Merzenich and White, 1977). Figure II.B.12 shows an aligned UCSF electrode array on the scale of the human implant. Each electrode of each dipole pair is located on the same radial line. This configuration has not been used in patients but, as just mentioned, has been used extensively in cat experiments. Figure II.B.13 shows the potential distributions calculated by the model for the aligned array of Figure II.B.12 with electrodes [3] and [4] being driven. The effective stimulus field is derived as described above, and is displayed in the top panel. Superimposed on the effective stimulus field is the curve of exponential falloff in the response fields measured for "well-positioned" electrodes in the cat (space constant = .87 mm.; see Merzenich and White (1977) for details). The agreement of the data from the model with the experimental results is truly remarkable, suggesting that the present approach to modeling is justified at this level of analysis. However, additional comparisons must be made to verify the model's accuracy for other situations and, more generally, direct

ALIGNED BIPOLAR
UCSF CONFIGURATION

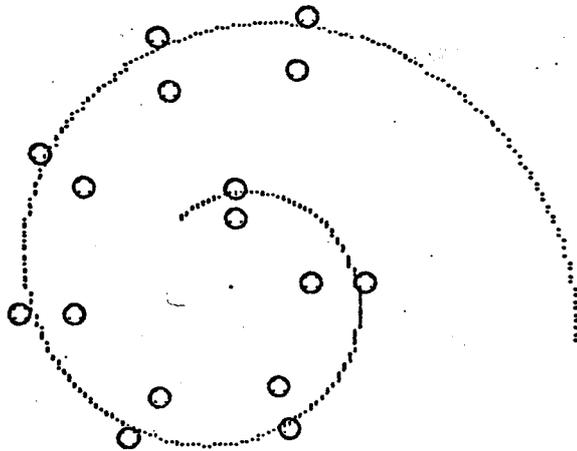


FIGURE II. B, 12

ALIGNED BIPOLAR
CONFIGURATION
(3-4)

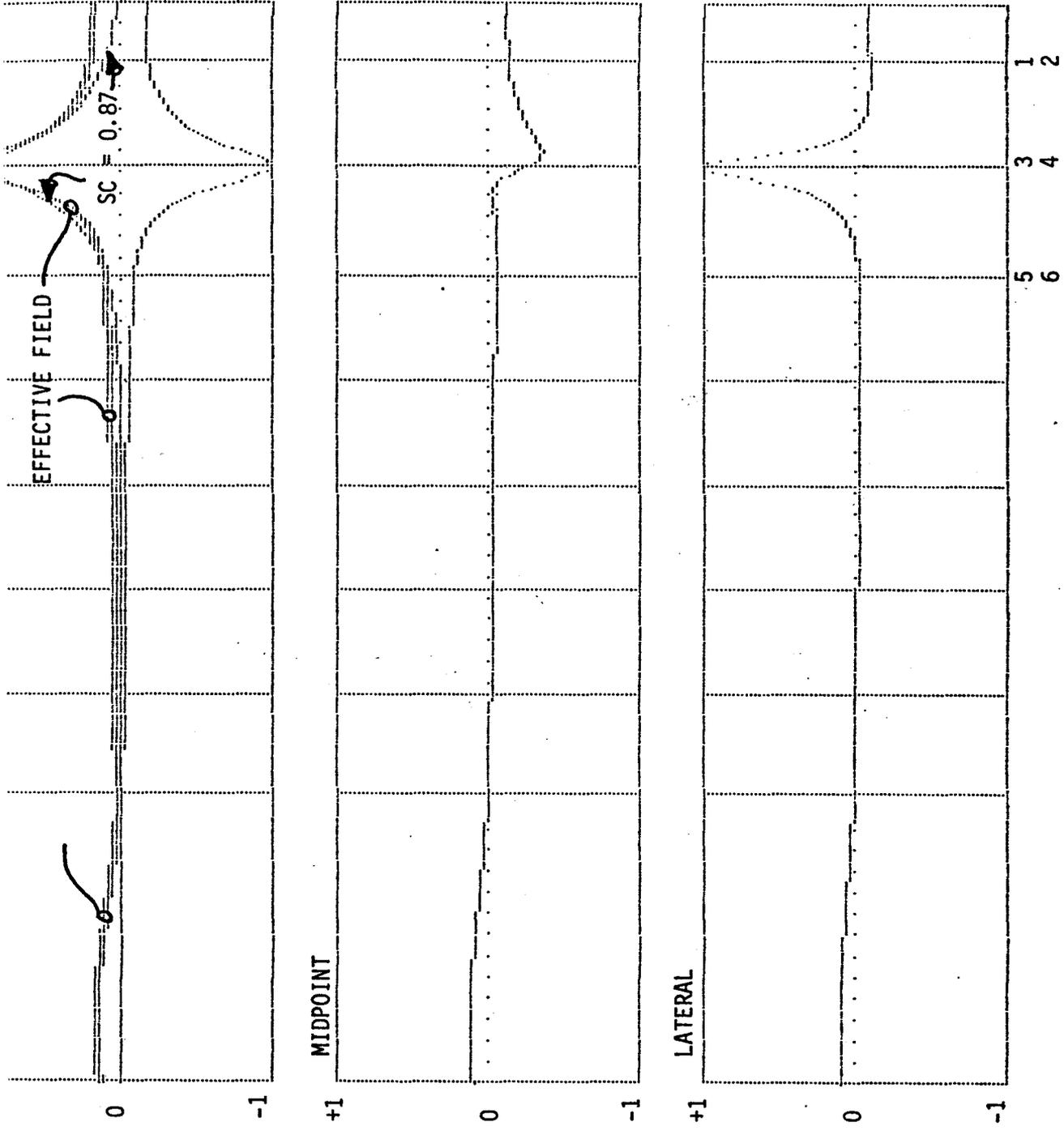


FIGURE II.8.13

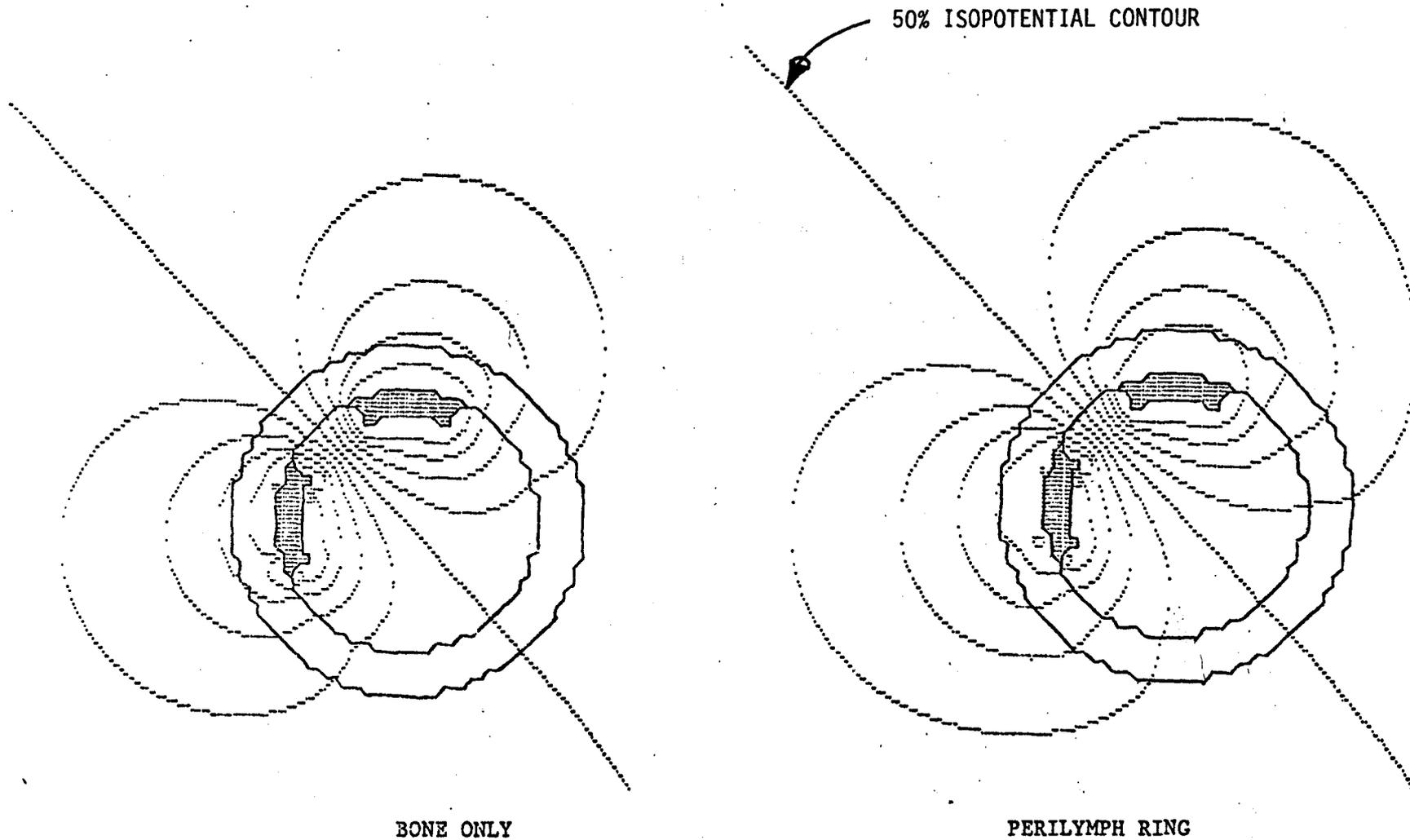
evaluation of the assumptions in the model should be made before we accept its predictions as valid. A direct evaluation of tissue-impedance effects (and the related assumption of uniform tissue properties) is therefore presented in the following subsection.

3. Tissue Impedance Considerations

Figure II.B.14 shows two depictions of the aligned UCSF electrode array in cross-section. (Refer to section II.B.2 for a more complete description of the computations involved in the cross-sectional model.) A concentric ring which surrounds the silastic insulator and electrodes has been added to the model. This ring represents the interface between the perilymph, which surrounds the electrode array in the scala tympani, and the bony wall of scala tympani. In the figure on the left, the ring's characteristics have been to represent bone beginning immediately at the surface of the silastic carrier and the electrodes. This condition corresponds to the assumptions of all the model calculations reported above. On the right is the same computation, but with the perilymph ring given the characteristics of perilymph. A slight expansion of the isopotential contours can be observed when the perilymph ring is included.

Figure II.B.15 shows lines of constant magnitude of current flux for the same two conditions described above for Figure II.B.14. Isocurrent magnitude contours (not flux lines) are shown at 5% intervals between the peak current magnitude flowing between the electrodes and zero. Note that with the perilymph ring included the isocurrent contours are more tightly constrained and do not spread significantly into the bony tissue. It is important to observe here, however, that the difference between the two conditions is relatively minor. At best, only about 10 % of the total current flux flows out to where the bony interface would lie in either set of conditions. In fact, the bulk of the current flows immediately along the surface of the silastic insulator between the two electrodes.

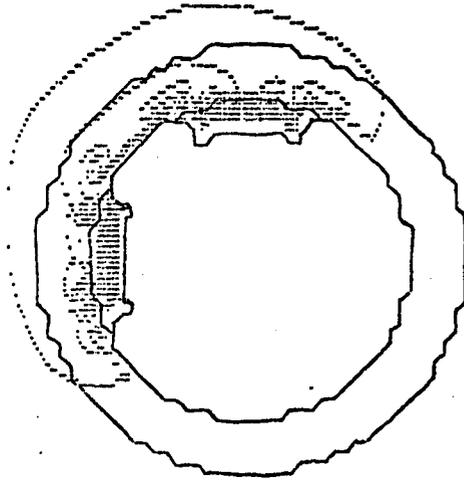
Figure II.B.16 gives a better perspective on the pattern of current flow between the electrodes, showing the current magnitude (normalized to the peak magnitude value) that flows normal to the 50% isopotential contour which between the two electrodes. Current magnitude is shown for both bone-only and perilymph-ring conditions. As can be seen, the magnitude of the



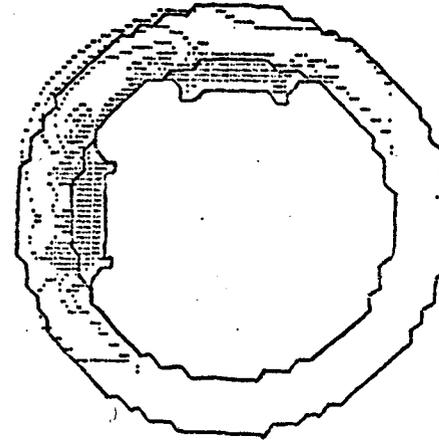
10 % ISOPOTENTIAL CONTOURS FOR ALIGNED UCSF BIPOLAR ELECTRODES

Resistivities:
 Insulator - 10^7 ohm-cm
 Bone - 800 ohm-cm
 Perilymph - 300 ohm-cm

FIGURE II. B. 14



BONE ONLY



PERILYMPH RING

5 % Constant Magnitude Current Lines

FIGURE II. 3. 15

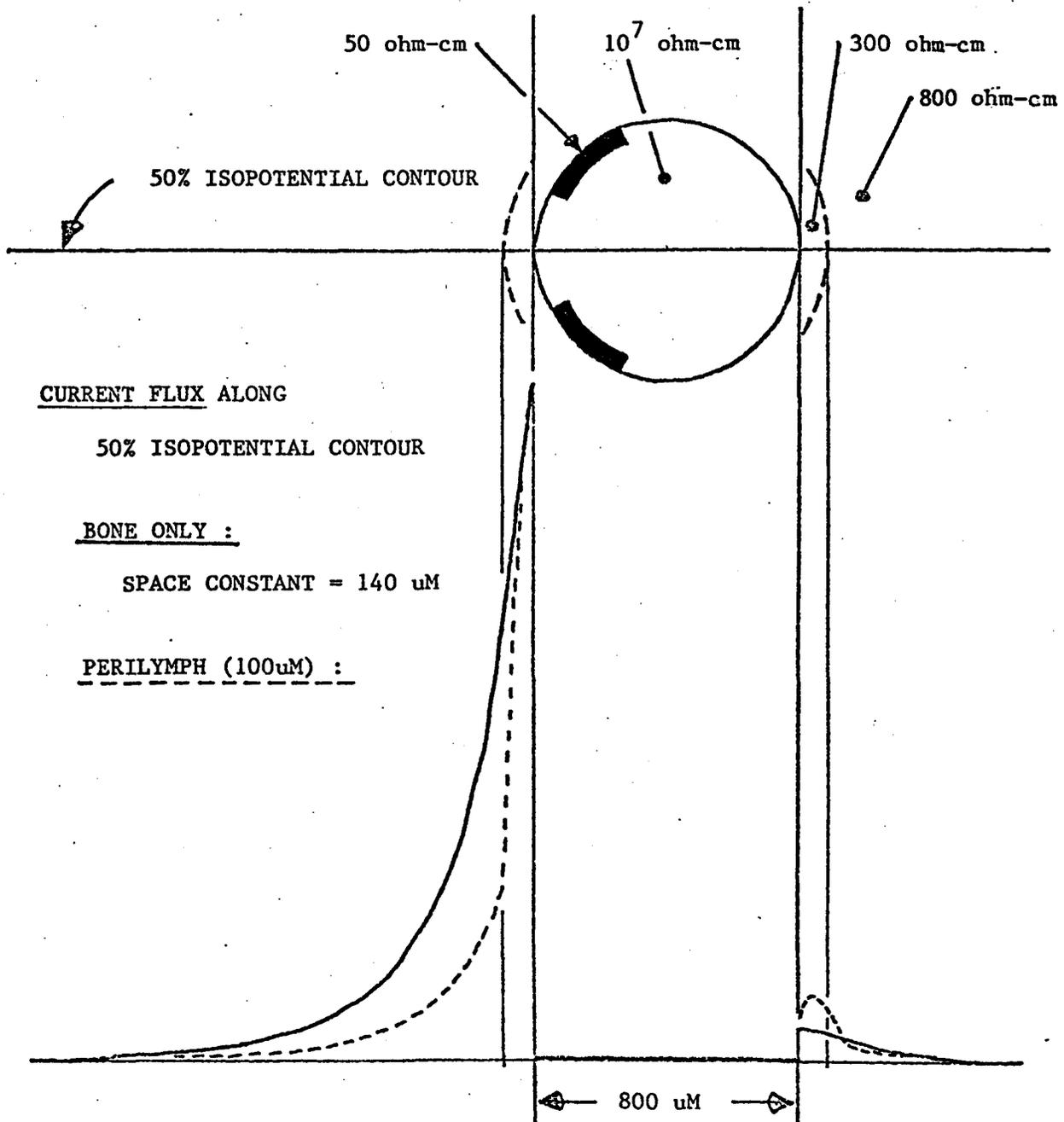


FIGURE II.3.16

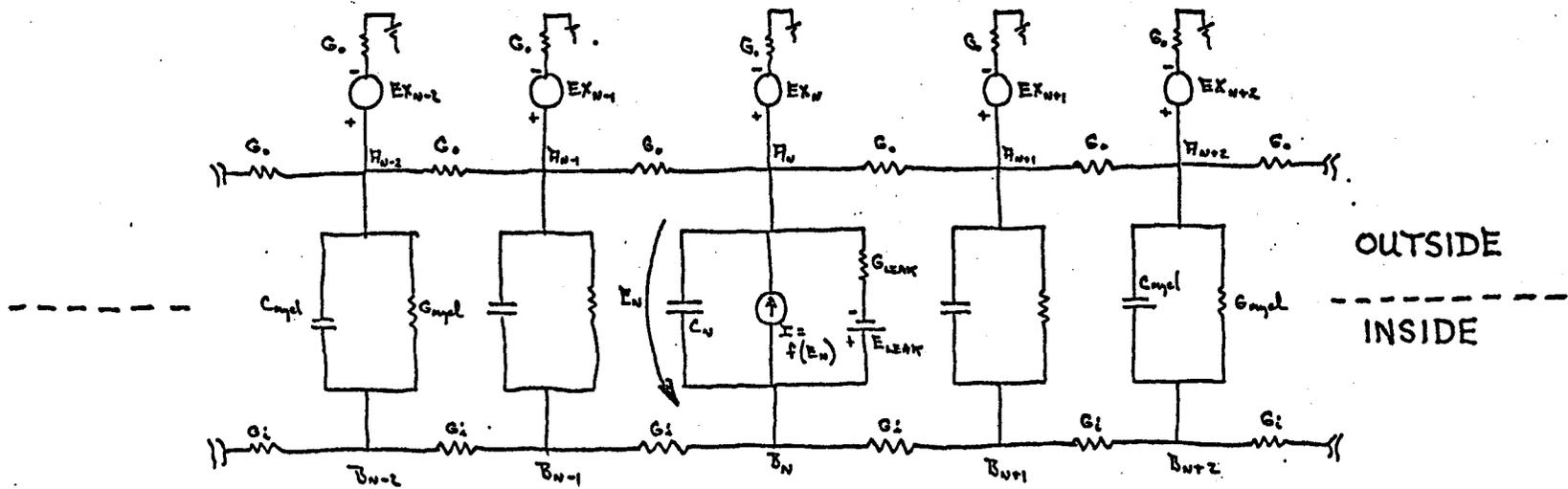
current drops rapidly as a function of distance from the silastic carrier. Little current flows on the far side of the silastic carrier, away from the electrodes. The present model assumes a 100 micron wide perilymph ring. In the practical application of an implant, placement of the electrodes within 100 microns of the bony wall would be extremely good placement with much greater distances expected typically. This would clearly be the case with electrode arrays other than the UCSF array, which do not have mechanical memories. Consequently, it is expected that relatively-small amounts of the injected stimulation current actually pass through the bony tissues. Rather, most of the current is shunted through the perilymph. In this case, we believe that simple modeling of the tissue impedances as being homogenous is a reasonable, first approximation to make.

One point to emphasize, however, is that this approximation is probably only valid in the case of bipolar stimulation. In the instance of monopolar stimulation to a remote reference, the bulk of the current flux leaving the electrode must necessarily flow across tissue boundaries. In this case, we believe that tissue impedances will play a substantial role in the behavior of the system. Further refinement of the present model will be required to accurately describe monopolar stimulation.

4. Myelinated Axon Model

The second half of the integrated field-neuron model is a lumped-element model of a myelinated axon. The present axon model extends the work of McNeal (1976) by providing for realistic impedance characterization of myelin, by incorporating mammalian node characteristics, and by allowing geometric scaling to the dimensions of cochlear neurons. This expansion allows computation of neural responses other than simple prethreshold responses, enables observation of propagation of spike activity, and permits study of complex, arbitrary stimuli delivered by intracochlear electrodes over extended periods.

A portion of this model is represented in Figure II.B.17. The neuron model comprises a section of a myelinated axon containing 19 active nodes of Ranvier. Each node is isolated from adjacent nodes by 9 myelinated segments. The ends of the neuron consist of 9 myelin segments, terminating in a sealed membrane. In all, 199 computational segments are included in the model. Nodes of Ranvier are located at segments 10, 20, ... , 180, 190. One internodal section includes characteristics of a cell body resembling the cell bodies of the bipolar cells of the spiral ganglion. Figure II.B.17 shows the electrical analogues assumed for each segment of the neuron. Five segments are shown, four myelin segments and one node of Ranvier. The myelin segments are assumed to be purely passive and comprise simply a parallel combination of the myelinated-segment transmembrane capacity and resistivity. The nodes of Ranvier are assumed to be nonlinear and are described by the Frankenhauser-Huxley node equations for the frog. Each node is described electrically by a transmembrane capacity in parallel with both a voltage-controlled current source and a series combination of a leakage resistance and battery source. The behavior of the voltage-controlled current source is described by Hodgkin-Huxley type equations adjusted for mammalian node characteristics (Chiu, Ritchie, Rogart and Stagg, 1979; Chiu, 1980; Brismar, 1980). The battery sources are adjusted to provide the normal resting potential for the axon at each of the active nodes. External to the axon at each segment is a voltage source, which drives through the impedance of the extracellular medium. These external voltage sources are set to the potential levels calculated by the finite-difference model.



Axon model showing a single Frankenhauser-Huxley node surrounded by four myelinated segments. A_N is the external segment voltage for axon segment N . B_N is the internal segment voltage. EX_N is the external voltage source for stimulation of that segment. See text for explanation.

Figure II.B.17

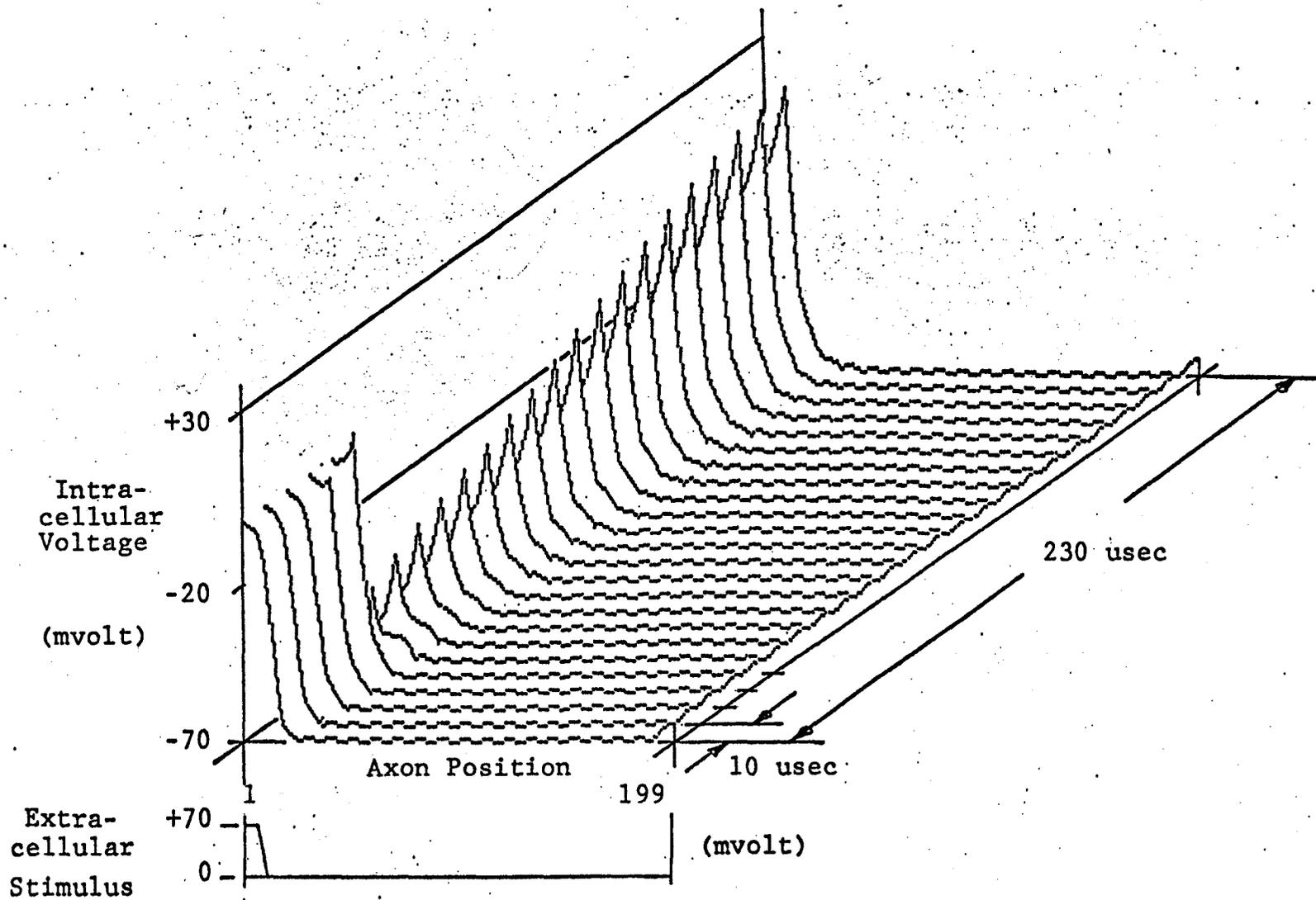
For initial validation of the model, parameters have been selected to describe the physical attributes of the myelinated, frog axon. This allows confirmation of the predictions of the model against a broad and well founded experimental literature. Once the model has been validated, parameters will be selected to more accurately describe the actual situation within the mammalian cochlea (Liberman and Oliver, 1984).

Figure II.B.18 illustrates the response of the model to a simple voltage pulse applied to the end node of the model axon. Specifically, the external voltage stimulus is a +70 mvolt, 50 microsecond stimulus with the profile along the axon as shown in the inset of Figure II.B.18. The display is a perspective view of sequential time samples of the internal voltages of the axon at all 199 computational segments. The abscissa is computational segment position along the axon, ranging from 1 to 199. The ordinate is the intracellular voltage referenced to the normal resting potential of -70 mvolts. Sequential time intervals recede into the background at 10 microsecond intervals. A total time span of 240 microseconds is displayed, beginning with the stimulus onset. As can be seen in Figure II.B.18, the 50 microsecond extracellular stimulus depolarizes the cell in the vicinity of the first node of Ranvier (segment 10). A slight regenerative depolarization can be seen at the node position at 40 and 50 microseconds into the stimulus. Then as the stimulus is turned off, the intracellular potential shifts again toward the resting potential. However, the first node is well into its regenerative phase and continues to depolarize the axon. From 60 to 240 microseconds in time, the early rising phase of an action potential is observed.

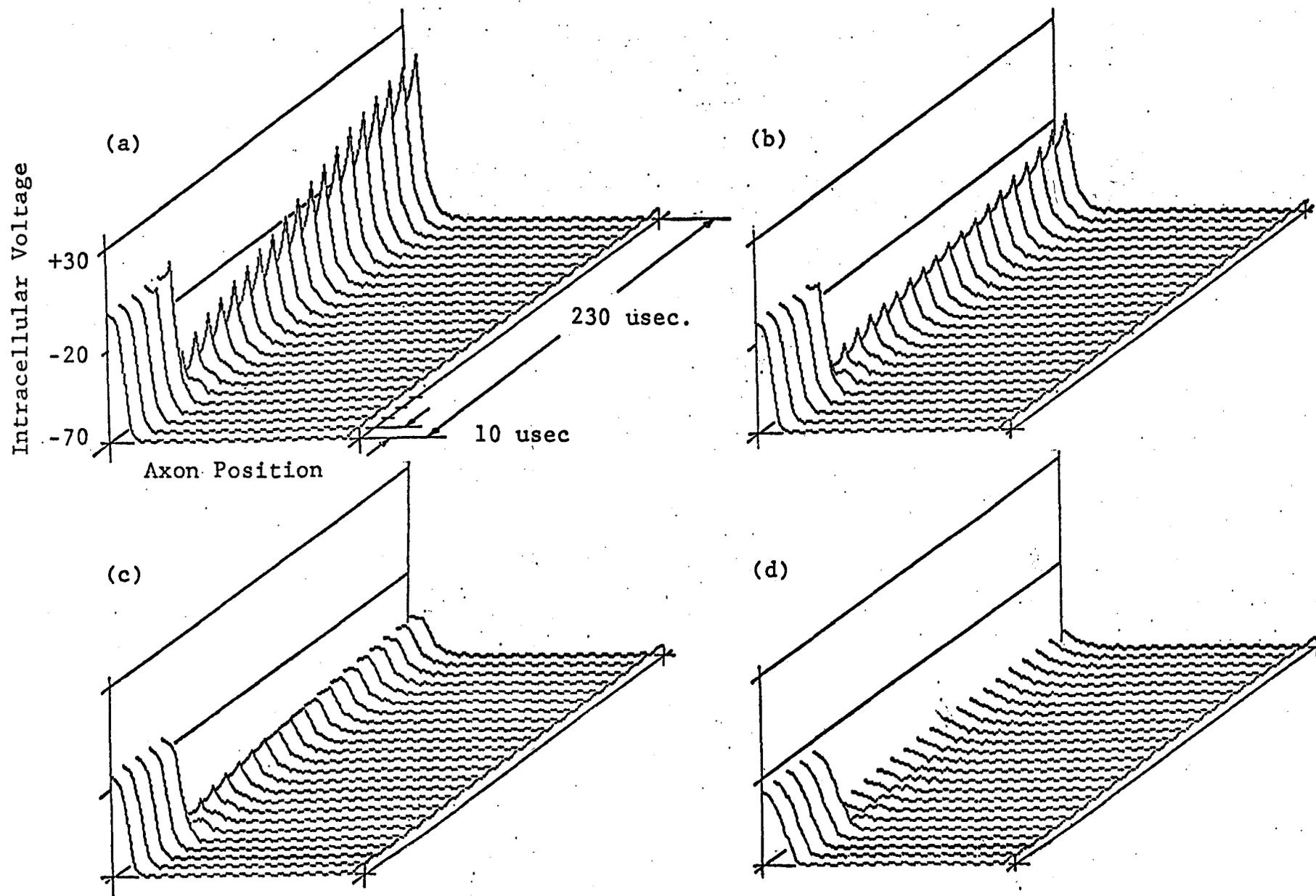
Consistent with normal axon behavior, graded stimuli reveal sub- and supra-threshold behavior of the model. Figures II.B.19 (a-d) show similar perspective displays of the model's response to the similar stimuli, varying only in their amplitude. Panel (a) is the response to a 70 mvolt stimulus and is identical to the response shown in Figure II.B.18. Panels (b), (c), and (d) show responses to 60, 55 and 40 mvolt stimuli, respectively. Spike generation is evident in panels (a) and (b). The response shown in panel (c) ultimately produces a spike at long latency if time is extended. The panel (d) response is clearly subthreshold.

As computation times are extended, more of the axon response may be observed. Figure II.B.20 shows a response display similar to the ones

Figure II, B. 18

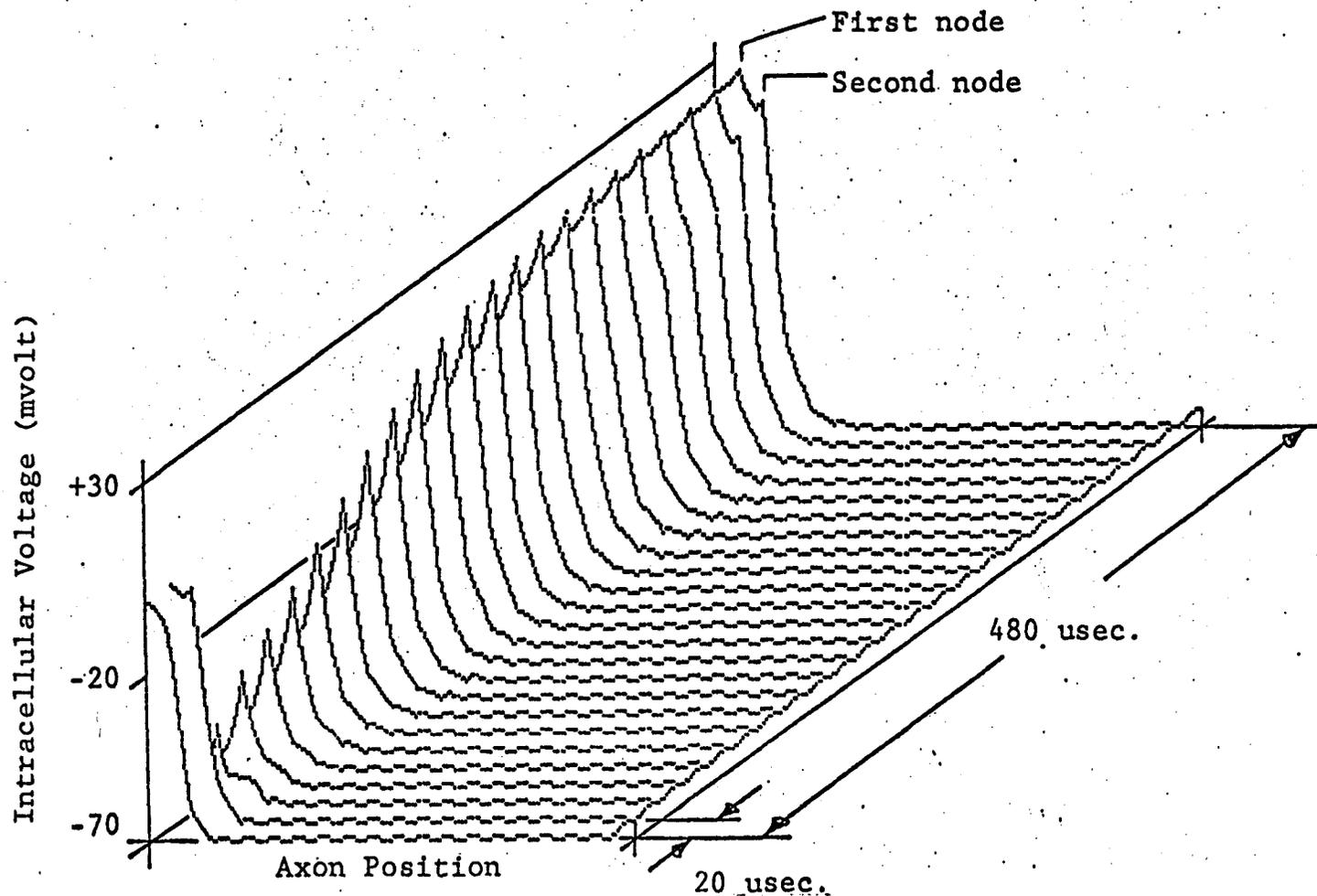


Perspective display of intracellular voltages along a 19 node axon model in response to a 50 usec. external voltage pulse of 70 mvolts. Extracellular stimulus profile along axon is shown in the lower panel. See text for full explanation.



Perspective displays of intracellular voltages along ^{the} axon model
 in response to 50 usec. external voltage pulses.
 (a) 70 mvolt stim; (b) 60 mvolt stim; (c) 55 mvolt stim; (d) 40 mvolt stim.

Figure II.B.19



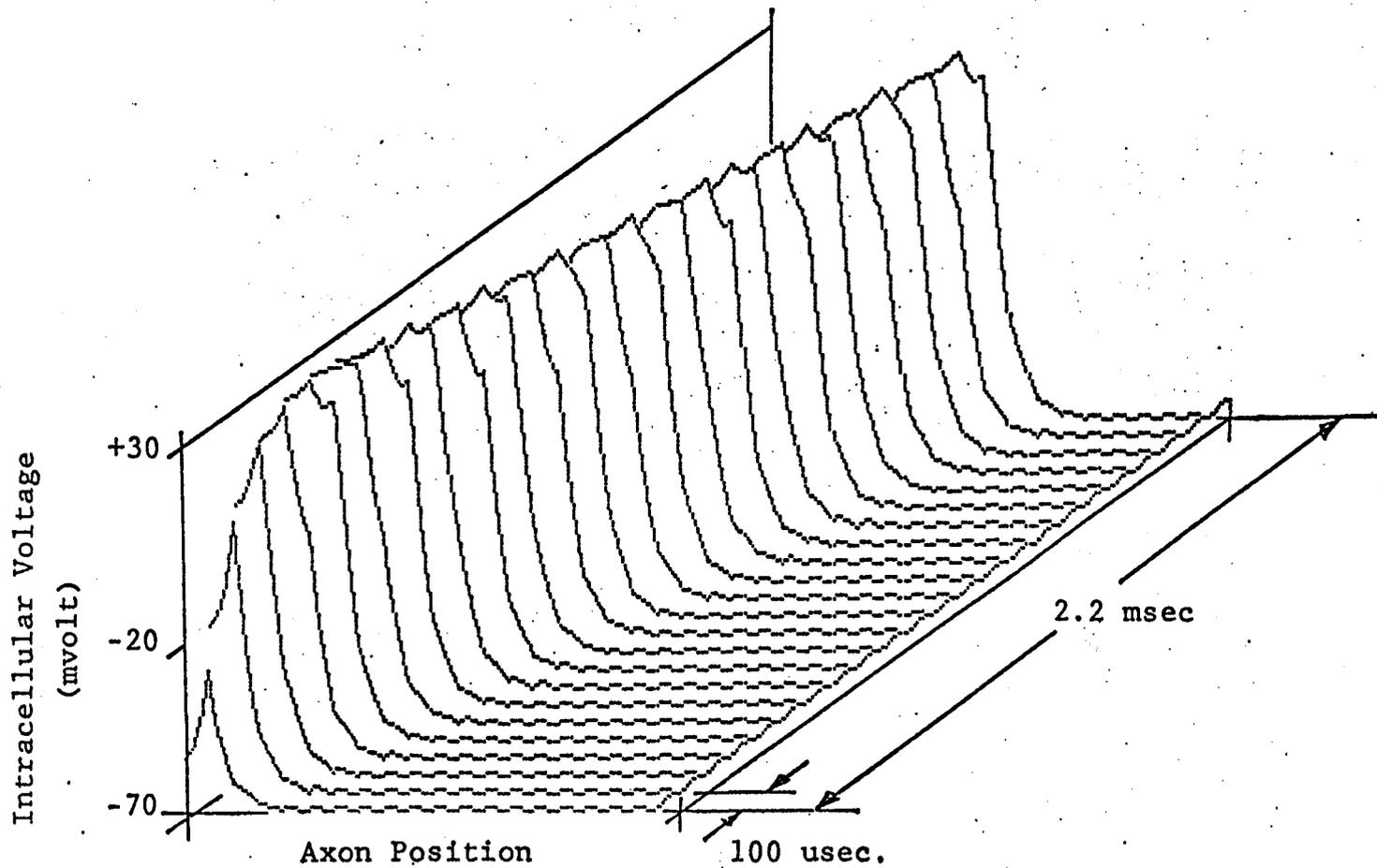
Perspective display of intracellular voltages produced under the same conditions as those in Figure III.2. In the present display, however, the time increment has been doubled to 20 usec. Note recruitment of the second node to begin propagation of the spike wavefront.

Figure II.B.20

previously discussed, however with time increments of 20.0 microseconds. As time progresses, a gradual spreading of the axonal depolarization begins, ultimately causing the second node of Ranvier to depolarize regeneratively.

Further expansion of the time display intervals to 100 microseconds reveals a spike wave front which has achieved a maximum potential level and clearly propagates along the axon, as shown in Figure II.B.21. Rough calculation indicates the propagation velocity of the spike front to be approximately 10 meters/second. The apparent discontinuities of the intracellular potentials along the axon, especially at the spike wave front, are typical of the so-called "saltatory conduction" of the spike from node to node (see Fitzhugh, 1962).

The axon model is presently being implemented on an Applied Dynamics AD10 numerical integrator at the National Biomedical Simulation Resource at Duke University. When installation is completed, neural model computations may be made in approximately twenty times real-time. This will provide an extremely powerful tool with which initial evaluations of speech processor designs may be made without patient involvement.



Perspective display of intracellular voltage along ^{the} axon model showing _A propagating spike following spike initiation.

Figure II, B, 21

5. Ensemble Model of Multiple Neurons

Speech encoding by electrical stimulation in the cochlea requires successful temporal control of an ensemble of neurons spatially distributed along the cochlear partition. A third model has been developed to describe ensemble responses of multiple neurons to stimulation by multiple electrodes. This model allows manipulation of field patterns and channel configurations of the electrodes, along with response characteristics and survival patterns of the neural elements.

In lieu of an expansive discussion of the details of this model here, the reader is referred to section III.A, Design of Stimulus Primitives, where a detailed example is presented.

6. Validation of the Models

Validation of the model is of crucial importance to its ultimate utility. Clearly, one validation approach is to predict the results of numerous animal studies of VIIIth nerve responses to intracochlear stimulation. This will be the most robust approach to validate the basic assumptions and techniques used in the models. Several examples have been presented in the preceding discussion where calculated model predictions agreed well with measured data from animal experiments. The abundance of published literature on VIIIth nerve recordings with speech stimuli will provide an initial database for comparison and prediction.

The RTI team is currently planning a joint effort with the UCSF group to perform specific animal experiments to evaluate our ability to control VIIIth nerve firing. Separate funding for this effort, independent of the present proposal, will be sought.

7. Studies of Intracochlear Evoked Responses

The RTI Patient Stimulator has been constructed with the capability to measure intracochlear potentials via nonstimulated electrodes within an implanted electrode array. These measurements can be made only ^{when} ~~with~~ the percutaneous cable is in use. The hardware for measuring these potentials is described in section II.A.3. Two classes of intracochlear potentials

will be recorded.

One is the recording of stimulus artifacts to subthreshold stimulation. This measurement provides information on the electrical field patterns within the implanted human cochlea without competing neural responses. In conjunction with the field mapping tools already described, estimates of the stimulus fields generated by the electrodes can be made. Consequently, ~~more~~ direct comparisons among various electrode configurations (monopolar vs bipolar, etc) can be made in the implanted situation. These comparisons may be very useful since the complications of patient subjective reporting and unknown neural survival patterns will be removed.

The other class of intracochlear potential is the neural evoked response. Recording of activity from neural elements within the cochlea will be a tremendous adjunct to the present methods of monitoring responses to stimulation in humans. These methods are the brainstem evoked response and the reporting of percepts by the patient. It appears that both of these measures may be biased when using the results to make inferences about the intracochlear mechanisms mediating electrically-evoked hearing. Both techniques involve substantial processing of the activity of the VIIIth nerve before a quantifiable response is produced. Intracochlear potential recording will provide a more direct measure of the specific gross activity of the spiral ganglion and the neural elements passing through the modiolus. This information may be available at a higher signal-to-noise ratio, allowing reduced averaging and test time. Interpretation of these data in light of the modeling results will produce a physiologically-based model of how the prosthesis interfaces to the nervous system. Many models exist for synthesizing compound action potentials based on the occurrence of neural spikes. Since the final output of the integrated field-neuron model is the spatial and temporal patterns of neural discharge evoked by a given stimulus, compound action potentials, associated with particular stimulus waveforms, can be synthesized for direct comparison with the measured compound potentials from the patients. Comparisons of observed data with data predicted by the model will allow evaluation of the model's success in describing the underlying mechanisms of stimulation. Knowledge of the physiological mechanisms of electrical stimulation combined with the information output from advanced speech processors, could, in turn, provide a basis for optimizing the stimulation strategy. Unique strategies, optimized for individual patients, may also be possible.

8. Application of the Model to Speech-Processor Design

The electrode array is the funnel through which the speech processor must push information into the central auditory system. Therefore, full knowledge of the characteristics of this funnel, in terms of the neural discharge patterns evoked by stimuli delivered to the electrode array, is essential for the rational design and further development of multichannel speech processors.

At the lowest level, the integrated field-neuron model and the ensemble model will help us define "stimulus primitives" that are effective in

- (1) controlling the temporal and spatial profiles of neural discharge around an electrode or electrode pair;
- (2) controlling or exploiting field interactions between electrodes; and
- (3) controlling for deleterious effects of pathology such as those produced by loss of spiral ganglion dendrites.

These stimulus primitives provide the translation functions a speech processor must employ in order to code speech features for maximum presentation of information to the central nervous system and for maximum fidelity of perception by the patient.

At another level, use of the model will help us to interpret intracochlear evoked responses. The results of these measurements and interpretations will be used to determine the extent and pattern of neuron survival around each pair of electrodes in the implanted array. This information, in turn, can be used as a powerful guide to selecting the optimal speech-processing strategy for the patient under test.

Finally, at the highest level, simulation of neural discharge patterns evoked by processed speech stimuli will be a powerful tool for evaluating the potential coding efficacies of speech-processing strategies. That is, with the model we can closely approximate the differences in discharge patterns produced between different speech tokens for each strategy we

evaluate. The objective here, of course, is to maximize these differences for a given class of pathology and for a given electrode design.

C. Computer-Based Simulator of Speech Processors for Auditory Prostheses and Related Software

In this section we will briefly describe software we have developed to (1) simulate speech processors for both single- and multichannel auditory prostheses; (2) generate the stimuli and implement the testing algorithms for obtaining basic measures of psychophysical performance with implant patients; and (3) sequence or randomize processed tokens and implement the testing algorithms for obtaining measures of speech reception and intelligibility with different speech-coding strategies. The computer system used for executing this software is described in section II.A of this proposal, as is the hardware interface that provides a safe, high-bandwidth communications link between the computer and implanted electrodes.

1. Block-Diagram Compiler

The problem of evaluating speech processors for auditory prostheses is not much different from the problem faced years ago by researchers at Bell Laboratories in their effort to design speech processors for reducing the bandwidth of telephone transmissions. As with auditory prostheses, there was no shortage of reasonable hypotheses worthy of careful evaluation. Two of the major difficulties were in assembling, sometimes with only minor modifications, the large number of candidate speech processors for evaluation, and then in documenting the results obtained with each one. The "turnaround time" for testing was prohibitive, and new methods were sought for paring this time down to less than 1 day and for improving the reliability of the tests. The brilliant solution of the Bell Labs researchers was to emulate in software the functional units of their speech processors in a "block-diagram" format familiar to electrical engineers. Using this system, the "block-diagram compiler," an investigator would specify the functions of individual blocks in an array of blocks and the topology of interconnections between the blocks (Golden, 1966; Kelly *et al.*, 1961). The functions included amplifiers, delay lines, limiters, rectifiers, multipliers, dividers, summers, and sine and cosine generators. Waveforms at any node in the simulated system could be displayed and copied to document performance, and selected nodes could be used for inputs and outputs. The inputs were usually sampled representations of speech read

from magnetic tape or disk storage and the outputs were most often extracted parameters of these inputs. Indeed, the block-diagram compiler found its greatest use in the early development of vocoder technology (Flanagan, 1972; Flanagan and Golden, 1966).

In our present project with the Neural Prosthesis Program, we have used the concept of the block-diagram compiler to build a computer-based system for the rapid and flexible simulation of speech processors for auditory prostheses. This new tool, developed with modern software techniques and utilizing modern digital signal processing (DSP) and speech analysis algorithms, is far more powerful than its predecessor and far more powerful in the present application than commercially-available DSP packages such as those offered by Signal Technology, Inc. (the "ILS" package) and others. 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 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;

*The following material in this subsection, with the exception of the last paragraph, is from our Fourth Quarterly Report of Progress for NIH contract N01-NS-2356.

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.

To illustrate the use and application of these programs, we will present an example of the specification and simulation of a relatively-simple processor. The principles of specification and simulation of more-complex systems are no different from the principles indicated in the following description. Therefore we hope this description will indicate to the reader the power of the simulation system for implementing in software any of the speech processors used in current auditory prostheses.

A session in which one or more of the above programs is to be used is initiated by calling program CPEXEC. This program presents the menu of options shown in Fig. II.2.1. With the exception of PREPARE, these options provide choices for calling the remaining programs in the set listed above. To specify the design of a new signal-processing system, then, the investigator enters a "1" (for "DESIGN A NEW SYSTEM"), and responds to the initial queries of the DESIGN program, as shown in Fig. II.2.2. In this particular case design # 2 is selected (up to 9999 design files can be stored on the disk) and the system sampling frequency is set at 20 kHz.

Once these preliminary entries have been made, the investigator is in a position to define the network of the signal-processing system on a block-

```
SELECT THE NEXT TASK FROM THE FOLLOWING OPTIONS:-
1 = DESIGN A NEW SYSTEM
2 = MODIFY AN EXISTING SYSTEM
3 = EXECUTE SIMULATION OF AN EXISTING SYSTEM
4 = DISPLAY OUTPUTS WRITTEN TO THE DISK DURING SYSTEM SIMULATION
5 = ASSIGN ELECTRODES TO RECEIVE DATA FROM OUTPUT FILES
6 = SAMPLE SPEECH OR OTHER DATA WITH THE A/D CONVERTER
7 = TEST A PATIENT BY SENDING DATA OUT OVER ASSIGNED ELECTRODE CHANNELS

ENTER > _
```

Fig. II.2.1. Menu presented by CPEXEC.

```
SELECT THE NEXT TASK FROM THE FOLLOWING OPTIONS:
1 = DESIGN A NEW SYSTEM
2 = MODIFY AN EXISTING SYSTEM
3 = EXECUTE SIMULATION OF AN EXISTING SYSTEM
4 = DISPLAY OUTPUTS WRITTEN TO THE DISK DURING SYSTEM SIMULATION
5 = ASSIGN ELECTRODES TO RECEIVE DATA FROM OUTPUT FILES
6 = SAMPLE SPEECH OR OTHER DATA WITH THE A/D CONVERTER
7 = TEST A PATIENT BY SENDING DATA OUT OVER ASSIGNED ELECTRODE CHANNELS

ENTER > 1
BUILD A NEW SYSTEM OR REVISE AN OLD ONE? (B = 0; R = 1): 0
ENTER DESIGN NUMBER (1-9999): 2
FILES FOR THIS DESIGN ALREADY EXISTS:
DO YOU WANT TO CREATE A NEW FILE OR WRITE OVER THE OLD ONE? (C = 0; W = 1): 1
ENTER SAMPLING FREQUENCY FOR SYSTEM: 20000_
```

Fig. II.2.2. CPEXEC menu followed by initial queries and responses of program DESIGN.

by-block basis. The system to be defined in the present example is shown in Fig. II.2.3. It consists of four bandpass channels of processing in which an approximate measure of the rms energy in each band is obtained. This

TEST SYSTEM FOR BLOCK-DIAGRAM SIMULATOR

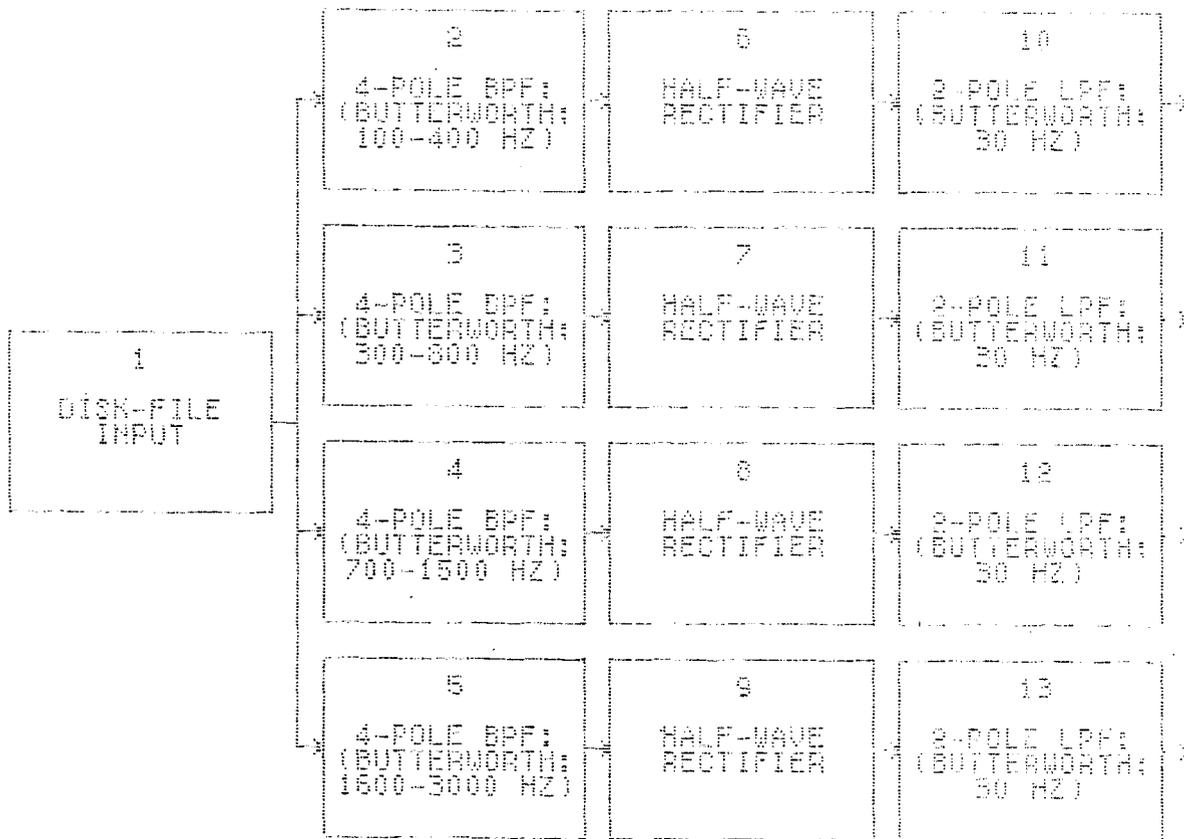


Fig. II.2.3. Test system for block-diagram simulator.

system is typical of the "front ends" of channel vocoders and could be used in somewhat modified form as the front end for certain types of speech processors for multichannel auditory prostheses.

The next screen presented after entering the global sampling frequency is the first of two main menus of the DESIGN program. These menus specify the functions that can be assigned to each block within the system's network of blocks. As indicated in Fig. II.2.4, these functions fall into general categories. The categories shown in the first menu are those for modules that implement standard functions of digital signal processing ("DSP"), speech analysis, generation of signals ("SIGNAL SOURCE"), and various mathematical operations. The range of choices within each category is

```

ENTER ONE OF THE FOLLOWING OPTIONS FOR THE FUNCTION OF BLOCK 1:
-----
MODULE CATEGORY  OPTION  FUNCTION
-----
DSP:             1 = FILTER
                2 = FFT ANALYZER
                3 = CEPSTRUM ANALYZER
                4 = DATA WINDOW
SPEECH ANALYSIS: 5 = LPC ANALYZER
                6 = FORMANT TRACKER
                7 = PITCH EXTRACTOR
SIGNAL SOURCE:   8 = NOISE GENERATOR
                9 = SIN/COS GENERATOR
               10 = PULSE-TRAIN GENERATOR
               11 = DISK FILE
MATH OPERATIONS: 12 = SUMMER
                13 = MULTIPLIER/INVERTER
                14 = DIVIDER
                15 = LOGARITHMIC CALCULATOR
                16 = INTEGRATOR
OTHER:           17 = SHOW REMAINING OPTIONS

ENTER OPTION: 11_

```

Fig. II.2.4. First main menu of the DESIGN program. The option entered at the bottom indicates that the function of block # 1 is to read information from the disk and to make this information available as a signal source to other blocks in the network.

designed to encompass most of the possibilities for speech processors for auditory prostheses. For cases in which other functions must be included, the second main menu (to be described below) provides an option for adding new functions to the list.

To begin specification of the signal-processing system shown in Fig. II.2.3, the function of block # 1 is entered. The number of this function is "11", which, once entered, calls a series of menus for specifying the source of information on the disk to be read by the block. The first of these menus is presented in Fig. II.2.5. This menu asks the investigator to indicate when in the simulation process he or she wants to name the disk file. The "@DATA" option allows the use of a generic file naming facility in the Data General AOS operating system.

In the example, a "2" is entered to select the disk input file at the time of system specification. This entry prompts the next menu and series of queries shown in Fig. II.2.6. The responses indicate that file "TUNA" is to be read by block # 1, and that this file does not have any of the special

```
SELECT DISK INPUT FILE FROM THE FOLLOWING OPTIONS:
  1 = @DATA
  2 = FILE TO BE SELECTED NOW
  3 = FILE TO BE SELECTED AT RUNTIME
ENTER >2_
```

Fig. II.2.5. First menu for full specification of the "disk file" option.

```
NOW SPECIFY FILE ATTRIBUTES:
  1 = FILE FROM RTI SPEECH DATABASE
  2 = FILE FROM DIGITIZED TOKENS OF MAC TEST
  3 = ILS FILE
  4 = OTHER
ENTER >4

ENTER FILENAME: TUNA
(NOTE THE THE PRESENT NUMBER OF INPUTS TO BE READ FROM THE DISK IS 1,
AND THAT THE MAXIMUM NUMBER IS 9)
IS THE OUTPUT OF THE PRESENT BLOCK TO BE WRITTEN TO THE DISK? (N = 0; Y = 1): 0_
```

Fig. II.2.6. Second menu and subsequent dialog for full specification of the "disk file" option.

headers used for files from the RTI speech data base, digitized tokens of the Minimal Auditory Capabilities Battery (the "MAC TEST"), or files generated by the Interactive Laboratory System of Signal Technology, Inc.

(option 3, the "ILS FILE"). If a response other than "4" is entered for

the choices indicated in the file attribute menu, the subsequent queries for identifying the file are tailored to the structure of the file headers. Finally, once the file is identified, an option is presented to write the output of the present block to the disk. This option appears at the end of the specification sequence for each block in the system network. In general, the outputs of all blocks that feed electrode channels must be written to the disk (for subsequent handling by programs ASNELEC and TEST), and other outputs the investigator wishes to examine after network simulation must also be written to the disk (for subsequent handling by program SHOWNTLL). The penalty associated with writing an excessive number of outputs to the disk is increased computer time required for simulation. Because file "TUNA" already exists on the disk, the option to write it back to the disk in another file is not taken for block # 1.

With the full specification of block # 1 as just described, the DESIGN program returns to its first main menu to read from the investigator the function of the next block in the network. The screen for the initial specification of block # 2 is shown in Fig. II.2.7. Reference to Fig. II.2.3 indicates that block # 2 is a bandpass filter, and therefore a "1" is entered in Fig. II.2.7 to specify the filter function. This action in turn calls up a subroutine for filter design that allows for flexible specification of classic IIR filters. The present set of options for filter design include the specification of (1) lowpass, highpass or bandpass response; (2) the class of filter response, where the choices include Butterworth, Chebychev and elliptic functions; (3) the break frequency or frequencies; and (4) a direct or indirect input of filter order. In the example network, block # 2 is a fourth-order Butterworth bandpass filter with break frequencies at 100 and 400 Hz. This filter is fully specified (and implemented) with the dialog shown in Fig. II.2.8. When responses have been entered for all the specification queries, the poles and zeros of the filter are computed and feedback is provided on its z-plane singularities. The typeout of singularities for the filter of block # 2 is shown in Fig. II.2.9. Next, a menu is presented for display of filter characteristics or for a return to the main program (for specification of another block). This menu is shown in Fig. II.2.10. As can be seen, the menu allows the investigator to call routines for the display of the steady-state, phase,

```

ENTER ONE OF THE FOLLOWING OPTIONS FOR THE FUNCTION OF BLOCK 2:

MODULE CATEGORY  OPTION  FUNCTION
-----
DSP:             1 =  FILTER
                2 =  FFT ANALYZER
                3 =  CEPSTRUM ANALYZER
                4 =  DATA WINDOW
SPEECH ANALYSIS: 5 =  LPC ANALYZER
                6 =  FORMANT TRACKER
                7 =  PITCH EXTRACTOR
SIGNAL SOURCE:   8 =  NOISE GENERATOR
                9 =  SIN/COS GENERATOR
               10 =  PULSE-TRAIN GENERATOR
                11 =  DISK FILE
MATH OPERATIONS: 12 =  SUMMER
                13 =  MULTIPLIER/INVERTER
                14 =  DIVIDER
                15 =  LOGARITHMIC CALCULATOR
                16 =  INTEGRATOR
OTHER:           17 =  SHOW REMAINING OPTIONS

ENTER OPTION: 1_

```

Fig. II.2.7. First main menu of the DESIGN program. The option entered at the bottom indicates that the function of block # 2 is to filter an input signal.

```

PLEASE SPECIFY THE FOLLOWING PARAMETERS OF FILTER DESIGN
ENTER FILTER TYPE (LOWPASS = 0; HIGHPASS = 1; BANDPASS = 2): 2
ENTER RESPONSE CLASS (BUTTERWORTH = 0; CHEBYCHEV = 1; ELLIPTIC = 2): 0
ENTER LOWER -3DB FREQUENCY (HZ): 100
ENTER UPPER -3DB FREQUENCY (HZ): 400
ENTER ORDER OF FILTER (MUST BE EVEN): 4
FINALLY, ENTER THE SAMPLING FREQUENCY (HZ): 20000

TIME TO COMPUTE POLES AND ZEROS:
ETIME = 0 SECONDS

SHOULD TYPEOUT OF FILTER PARAMETERS BE SENT TO THE LPT? (N = 0; Y = 1): 0_

```

Fig. II.2.8. Dialog for the full specification of a fourth-order bandpass filter with break frequencies at 100 and 400 Hz.

impulse or step responses of the filter. In addition, the filter design can

```

TIME TO COMPUTE POLES AND ZEROS:
ETIME =      0 SECONDS

SHOULD TYPEOUT OF FILTER PARAMETERS BE SENT TO THE LPT? (N = 0; Y = 1): 0

A 4TH-ORDER FILTER IS SPECIFIED WITH THE FOLLOWING Z-PLANE SINGULARITIES:

ZERO( 1) =  1.00000 + J .00000
ZERO( 2) = -1.00000 + J .00000
ZERO( 3) = -1.00000 + J .00000
ZERO( 4) =  1.00000 + J .00000

POLE( 1) =  .98336 - J .03126
POLE( 2) =  .94627 + J .09353
POLE( 3) =  .94627 - J .09353
POLE( 4) =  .98336 + J .03126

TIME TO DESIGN NETWORK:
ETIME =      1 SECONDS

PAUSE _

```

Fig. II.2.9. Typeout of z-plane singularities for the filter specified in Fig. II.2.8.

```

SELECT ONE OF THE FOLLOWING OPTIONS FOR DISPLAY OF FILTER CHARACTERISTICS:

1 = PLOT STEADY-STATE RESPONSE FROM Z-PLANE SINGULARITIES
2 = PLOT PHASE RESPONSE FROM Z-PLANE SINGULARITIES
3 = PLOT IMPULSE RESPONSE
4 = PLOT STEP RESPONSE
5 = DESIGN NEW FILTER
6 = RETURN TO BLOCK-DIAGRAM DESIGN PROGRAM

ENTER OPTION: 1
TIME TO CALCULATE FREQUENCY RESPONSE =
ETIME =      1 SECONDS
PLEASE ENTER # OF VERTICAL DISPLAY UNITS / DB (2, 3 OR 5): 5_

```

Fig. II.2.10. Menu of options for the display of filter characteristics or for a return to the main DESIGN program. The option taken, along with the subsequent dialog, calls for a plot of the filter's steady-state magnitude response.

be revised by taking option 5 to "DESIGN NEW FILTER." The highly

interactive nature of communication with the filter design subroutine provides the investigator with a powerful facility for specifying exactly the desired characteristics of filter response. This type of interactive communication is a general feature of the DESIGN program, and is used in the specification of all complex functions indicated in the main menus of Figs. II.2.4 and II.2.16.

A typical series of displays an investigator might ask the filter subroutine to produce is illustrated in Figs. II.2.11 through II.2.13. First, the responses to the queries shown at the bottom of Fig. II.2.10 indicate that a plot of the steady-state magnitude response is desired, and that the ordinate is to have 5 display units per dB of attenuation. The resulting plot is presented in Fig. II.2.11. A log-log graph of the response is drawn, so that the investigator can easily appreciate the rates of falloff in the skirts of the filter in terms of dB/decade of frequency. Finally, an option to copy the display for a permanent plot on the line printer is presented; this copy query is made for all graphs drawn by the DESIGN program.

Once the response to the copy query is entered (and the copy made if the response is a "1"), the menu for "display characteristics or return to the main program" is presented again, as shown in Fig. II.2.12. This allows the investigator to examine other characteristics of the filter response or to refine the filter design before returning to the main program. In Fig. II.2.12, the option to display the impulse response of the filter is taken. The dialog following the specification of the impulse response establishes various attributes of the display. For example, the time axis can be manipulated (i.e., compressed) by skipping points in the display. To assist the investigator in setting the amplification factor and y position of the trace, the program provides feedback on the maximum and minimum values in the calculated impulse response. When the responses to all queries presented in the dialog of Fig. II.2.12 have been entered, the impulse response is drawn, as shown in Fig. II.2.13. As before, a copy query is presented with the display so that a copy can be made if desired.

The final step in the specification of block # 2 is illustrated in Fig. II.2.14. This is the same menu presented in previous figures for "display characteristics or return to the main program," but in this case the option

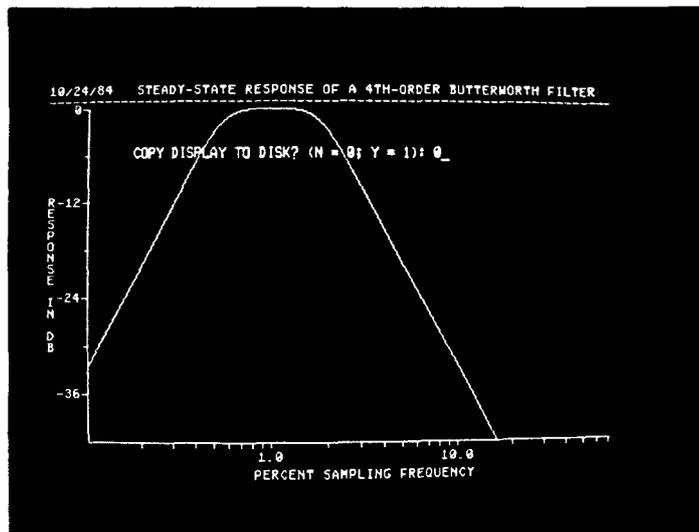


Fig. II.2.11. Plot of the steady-state magnitude response of the filter specified in Fig. II.2.8.

```

SELECT ONE OF THE FOLLOWING OPTIONS FOR DISPLAY OF FILTER CHARACTERISTICS:
1 = PLOT STEADY-STATE RESPONSE FROM Z-PLANE SINGULARITIES
2 = PLOT PHASE RESPONSE FROM Z-PLANE SINGULARITIES
3 = PLOT IMPULSE RESPONSE
4 = PLOT STEP RESPONSE
5 = DESIGN NEW FILTER
6 = RETURN TO BLOCK-DIAGRAM DESIGN PROGRAM

ENTER OPTION: 3
SKIP POINTS IN DISPLAY? (N = 0; Y = 1): 0
MAX VALUE IN IMPULSE RESPONSE = 3.69966E-02
MIN VALUE IN IMPULSE RESPONSE = -3.02497E-02
ENTER AMPLIFICATION FACTOR FOR DISPLAY: 3000
ENTER Y POSITION OF ZERO-LEVEL FOR TRACE: 123_

```

Fig. II.2.12. Menu of options for the display of filter characteristics or for a return to the main DESIGN program. The option taken, along with the subsequent dialog, calls for a plot of the filter's impulse response.

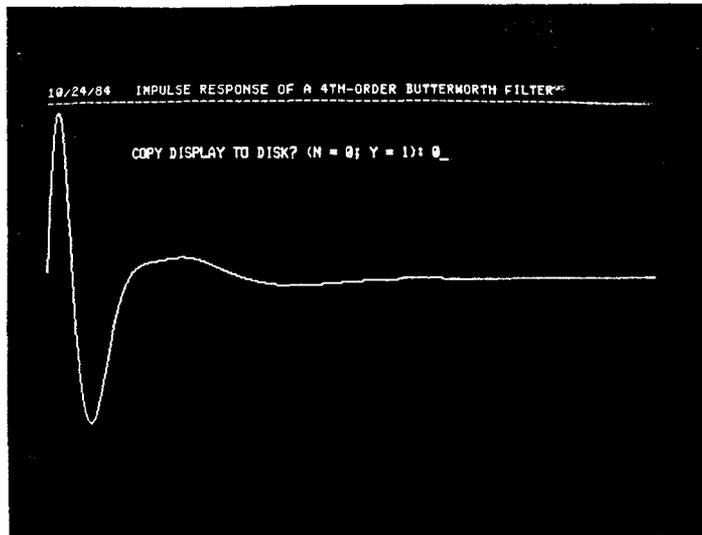


Fig. II.2.13. Plot of the impulse response of the filter specified in Fig. II.2.8.

```

SELECT ONE OF THE FOLLOWING OPTIONS FOR DISPLAY OF FILTER CHARACTERISTICS:
  1 = PLOT STEADY-STATE RESPONSE FROM Z-PLANE SINGULARITIES
  2 = PLOT PHASE RESPONSE FROM Z-PLANE SINGULARITIES
  3 = PLOT IMPULSE RESPONSE
  4 = PLOT STEP RESPONSE
  5 = DESIGN NEW FILTER
  6 = RETURN TO BLOCK-DIAGRAM DESIGN PROGRAM

ENTER OPTION: 6
INPUT BLOCK NUMBER = 1
IS THE OUTPUT OF THE PRESENT BLOCK TO BE WRITTEN TO THE DISK? (N = 0; Y = 1): 1

```

Fig. II.2.14. Menu of options for the display of filter characteristics or for a return to the main DESIGN program. The option taken, along with the subsequent dialog, calls for a return to the main DESIGN program. The dialog also specifies the input to the filter and establishes that the output of the present block is to be written to the disk during system simulation.

to return to the main program is taken. Unlike the signal source function

specified for block # 1, the filter must have an input from one of the outputs of another block in the network. For the system defined up to this point, the only possibility is the output of block # 1. Any other entry would produce an error message and the investigator would be asked again for the input block number. Valid entries of input block numbers establish the topology of the network in that they specify for the final system all interconnections between blocks. In general, blocks can be specified in any order (and not necessarily the left-to-right order of the present example) as long as an input can be identified for each block that requires one.

As mentioned before, the last query in the specification of all blocks is one whose response indicates whether the output (or outputs) of the present block is (or are) to be written to the disk. In the example of Fig. II.2.14, the response is affirmative and therefore the output of block # 2 will be available at a later time for examination with program SHOWNTELL, or for electrical stimulation with programs ASNELEC and TEST.

Once the filter for block # 2 is fully specified as described above, the first main menu of the DESIGN program appears for initial specification of the next block in the network. Because the procedure for specifying the filters of blocks 3, 4 and 5 (see Fig. II.2.3) is similar to the procedure just illustrated for block # 2, we will skip to the menu shown in Fig. II.2.15, which is the screen that appears after the specification of block # 5. Thus, the investigator is now in a position to specify the function of block # 6 in the network. Reference to Fig. II.2.3 indicates that the function of this block is to rectify the output of block # 2. However, rectification does not appear as an option in the first main menu of Fig. II.2.15, so the second main menu is called by entering a "17" to show the remaining options. The second main menu which then appears is shown in Fig. II.2.16. Options presented in this menu include those for specifying various circuit functions familiar to electrical engineers and for implementing various control functions of the DESIGN program. The control functions are listed under module category "OTHER", and include facilities to (1) read a subsystem for the present block from another design stored on the disk; (2) select a user-defined rule, the code of which is also stored on the disk; (3) identify a new user-defined rule to be added to the library of user-defined rules; (4) show the topology of the present system as

```

ENTER ONE OF THE FOLLOWING OPTIONS FOR THE FUNCTION OF BLOCK 6:
MODULE CATEGORY  OPTION  FUNCTION
-----
DSP:             1 =  FILTER
                2 =  FFT ANALYZER
                3 =  CEPSTRUM ANALYZER
                4 =  DATA WINDOW
SPEECH ANALYSIS: 5 =  LPC ANALYZER
                6 =  FORMANT TRACKER
                7 =  PITCH EXTRACTOR
SIGNAL SOURCE:   8 =  NOISE GENERATOR
                9 =  SIN/COS GENERATOR
               10 =  PULSE-TRAIN GENERATOR
               11 =  DISK FILE
MATH OPERATIONS: 12 =  SUMMER
               13 =  MULTIPLIER/INVERTER
               14 =  DIVIDER
               15 =  LOGARITHMIC CALCULATOR
               16 =  INTEGRATOR
OTHER:           17 =  SHOW REMAINING OPTIONS

ENTER OPTION: 17_

```

Fig. II.2.15. First main menu of the DESIGN program. The option entered at the bottom indicates that the function of block # 6 is not displayed on this menu and therefore a request is made to show the second main menu, presented below in Fig. II.2.16.

```

ENTER ONE OF THE FOLLOWING OPTIONS FOR THE FUNCTION OF BLOCK 6:
MODULE CATEGORY  OPTION  FUNCTION
-----
CIRCUIT FCMS:   18 =  COMPRESSOR
                19 =  ZERO-CROSSING COUNTER
                20 =  PEAK DETECTOR
                21 =  WINDOW COMPARATOR
                22 =  LEVEL COMPARATOR
                23 =  ONE SHOT (MONOSTABLE MULTIVIBRATOR)
                24 =  FLIP-FLOP
                25 =  SWITCH
                26 =  RECTIFIER
                27 =  UNIT DELAY OPERATOR
OTHER:           28 =  READ SUBSYSTEM FOR PRESENT BLOCK FROM ANOTHER DESIGN
                29 =  SELECT A USER-DEFINED RULE
                30 =  IDENTIFY A NEW USER-DEFINED RULE
                31 =  SHOW TOPOLOGY OF PRESENT SYSTEM
                32 =  RETURN TO PREVIOUS SCREEN
                33 =  REVISE A BLOCK
                34 =  EXIT FROM DESIGN PROGRAM

ENTER OPTION: 26_

```

Fig. II.2.16. Second main menu of the DESIGN program. The option entered at the bottom indicates that the function of block # 6 is to rectify an input signal.

defined by the investigator up to this point; (5) return to the previous

screen, the first main menu of the DESIGN program; (6) revise a previously-specified block; and (7) exit from the DESIGN program. As might be appreciated from the list just given, the first three control functions provide powerful tools for the specification of complex signal-processing systems. For example, the entire signal-processing system of Fig. II.2.3 could be read as a single block into a larger and more complex signal-processing system using the command to "READ SUBSYSTEM FOR PRESENT BLOCK FROM ANOTHER DESIGN." Because many signal-processing systems can be configured from a few functional subsystems (such as the front end of a four-channel vocoder, as illustrated here), this command can save an enormous amount of the investigator's time in building different processors.

The other two options that are useful in the specification of complex systems are those for selecting and identifying a user-defined rule. These options allow the investigator to build and use functions that are not included in the main menus of the DESIGN program (Figs. II.2.4 and II.2.16). An example of one such function would be that of a microprocessor embedded in a signal-processing system. Specification and simulation of the logic decisions made by a microprocessor are straightforward with user-defined rules; specification and simulation of these same decisions would be very difficult without user-defined rules.

To return now to the specification of the example system, we note that the option for specifying a rectifier is "26", and is therefore entered at the bottom of the menu in Fig. II.2.16. The queries presented after this entry has been made are indicated in Fig. II.2.17. As can be appreciated from the short list of questions, full specification of a rectifier is much simpler than full specification of a filter. In the example, a half-wave rectifier that processes floating-point values is specified (the choice between floating-point and integer values is usually made on the basis of processing speed and the expected range of the data). Finally, as in the specification of other blocks, the input to the rectifier is identified and an indication is given on whether the output of the present block is to be written to the disk.

The procedures used for specifying the remaining blocks in the network are similar or identical to those outlined above. Once all blocks have been specified, the design data are stored on the disk by invoking option "34" in

```
SHOULD THE RECTIFIER BE HALF WAVE OR FULL WAVE? (H = 0; F = 1): 0
SHOULD THIS BLOCK PROCESS INTEGER OR FLOATING-POINT VALUES? (I = 0; FP = 1): 1
INPUT BLOCK NUMBER = 2
IS THIS OUTPUT TO BE WRITTEN TO THE DISK? (N = 0; Y = 1): 1
```

Fig. II.2.17. Dialog for full specification of a half-wave rectifier that processes floating-point values.

the second main menu of the DESIGN program. This action is illustrated in Fig. II.2.18. The command to exit from the DESIGN program first ^{prompts} prompts the investigator to label the design with a title and then calls into memory the program PREPARE. The function of PREPARE is to arrange the information stored in the design files in such a way that the time required for subsequent simulation of the network by program EXECUTE is minimized. Fig. II.2.19 shows the screen that appears when PREPARE is doing its work. For a design of the complexity indicated in Fig. II.2.3 (the example system), PREPARE requires about 10 seconds of computer time to run.

When PREPARE is finished, CPEXEC is again invoked and the investigator can call any other program in the set listed at the beginning of this section. To simulate the example network just designed, for example, an entry of "3" would be made in response to the CPEXEC menu as shown in Fig. II.2.20. This entry calls program EXECUTE which first asks for the number of the design to be simulated and then asks whether the entire input file(s) or only of segment of the input file(s) should be processed. At various points in the simulation feedback is provided by EXECUTE to inform the investigator of processing status. The two queries and initial feedback from EXECUTE are shown in Fig. II.2.21.

```
ENTER ONE OF THE FOLLOWING OPTIONS FOR THE FUNCTION OF BLOCK 14:
MODULE CATEGORY  OPTION  FUNCTION
-----
CIRCUIT FCNS:   18 = COMPRESSOR
                 19 = ZERO-CROSSING COUNTER
                 20 = PEAK DETECTOR
                 21 = WINDOW COMPARATOR
                 22 = LEVEL COMPARATOR
                 23 = ONE SHOT (MONOSTABLE MULTIVIBRATOR)
                 24 = FLIP-FLOP
                 25 = SWITCH
                 26 = RECTIFIER
                 27 = UNIT DELAY OPERATOR

OTHER:          28 = READ SUBSYSTEM FOR PRESENT BLOCK FROM ANOTHER DESIGN
                 29 = SELECT A USER-DEFINED RULE
                 30 = IDENTIFY A NEW USER-DEFINED RULE
                 31 = SHOW TOPOLOGY OF PRESENT SYSTEM
                 32 = RETURN TO PREVIOUS SCREEN
                 33 = REVISE A BLOCK
                 34 = EXIT FROM DESIGN PROGRAM

ENTER OPTION: 34_
```

Fig. II.2.18. Second main menu of the DESIGN program. The option entered at the bottom indicates that the specification of the present design is complete, and that an exit should be taken from the DESIGN program.

```
TO CONCLUDE, PLEASE ENTER TITLE OF THIS DESIGN ON THE FOLLOWING LINE:
TEST SYSTEM 1
NOW PREPARING DESIGN FILES FOR USE BY THE SIMULATION PROGRAM
```

Fig. II.2.19. Screen that appears during the execution of program PREPARE.

```
SELECT THE NEXT TASK FROM THE FOLLOWING OPTIONS:
1 = DESIGN A NEW SYSTEM
2 = MODIFY AN EXISTING SYSTEM
3 = EXECUTE SIMULATION OF AN EXISTING SYSTEM
4 = DISPLAY OUTPUTS WRITTEN TO THE DISK DURING SYSTEM SIMULATION
5 = ASSIGN ELECTRODES TO RECEIVE DATA FROM OUTPUT FILES
6 = SAMPLE SPEECH OR OTHER DATA WITH THE A/D CONVERTER
7 = TEST A PATIENT BY SENDING DATA OUT OVER ASSIGNED ELECTRODE CHANNELS

ENTER ) 3_
```

Fig. II.2.20. Menu presented by CPEXEC. The option entered at the bottom indicates that the next program to be called is EXECUTE (for the simulation of a signal-processing system).

```
SELECT THE NEXT TASK FROM THE FOLLOWING OPTIONS:
1 = DESIGN A NEW SYSTEM
2 = MODIFY AN EXISTING SYSTEM
3 = EXECUTE SIMULATION OF AN EXISTING SYSTEM
4 = DISPLAY OUTPUTS WRITTEN TO THE DISK DURING SYSTEM SIMULATION
5 = ASSIGN ELECTRODES TO RECEIVE DATA FROM OUTPUT FILES
6 = SAMPLE SPEECH OR OTHER DATA WITH THE A/D CONVERTER
7 = TEST A PATIENT BY SENDING DATA OUT OVER ASSIGNED ELECTRODE CHANNELS

ENTER ) 3
ENTER DESIGN NUMBER (1-9999): 2
NOW READING INITIALIZATION DATA AND OPENING FILES
PROCESS ENTIRE INPUT FILE OR ONLY A PORTION OF IT? (E = 0; P = 1): 0
NOW SIMULATING NETWORK
SAMPLE NUMBER = 1024
```

Fig. II.2.21. CPEXEC menu followed by the initial dialog and feedback of program EXECUTE. The dialog indicates that design # 2 is to be simulated, and that the entire input file is to be processed.

Among the programs in the set, EXECUTE is by far the most complex in

terms of the computer resources it must handle. First, data files and system design files are all stored in the extended memory of the Eclipse to minimize disk input/output operations during simulation. Next, the logic controlling the order in which blocks are simulated, the length of data segments handled in each "chunk" of processing, and number of passes through feedback loops that lack memory elements is also carefully tailored to optimize the speed of execution. Finally, disk I/O operations that must be made at "chunk intervals" are coded for rapid transfer of data in 256-, 512- or 1024-word blocks, again to speed up the simulation of complex signal-processing systems.

As indicated in Fig. II.2.22, the total number of 1024-word pages used in extended memory for the example system is 14, and the total time required for processing 7168 input samples is 15 seconds. This time compares quite favorably with the performance of other simulation programs that are designed to run on general-purpose minicomputers. For example, simulation of the same system using the Interactive Laboratory System of Signal Technology, Inc. requires 280 seconds on the Eclipse. For a 10 kHz sampling rate, then, these figures correspond to 21 times real time for the RTI computer-based simulator and 391 times real time for the ILS simulator. The speed advantage of the RTI simulator is important for the present application in that it (1) allows for the evaluation of different processing strategies within single sessions with a patient and (2) allows for the processing of all tokens in the "miniMAC" test by several different processing strategies in overnight runs.

Once simulation of the network is completed, program EXECUTE closes open files on the disk and then pauses so that the investigator can examine the feedback provided during the simulation (see Fig. II.2.22). When the investigator is ready to proceed, he or she strikes any key on the terminal and program CPEXEC is called. The menu for selecting the next task is then presented once again, as illustrated in Fig. II.2.23. To display the output signals written to the disk during the previous simulation, an entry of "4" is made in Fig. II.2.23 to call program SHOWNTELL. Figs. II.2.24 through II.2.28 show typical displays drawn by this program. The first display is of the input disk file "TUNA". As is evident from the waveform (see Fig. II.2.24), file TUNA is a linear frequency sweep, the limits of which

```

2 = MODIFY AN EXISTING SYSTEM
3 = EXECUTE SIMULATION OF AN EXISTING SYSTEM
4 = DISPLAY OUTPUTS WRITTEN TO THE DISK DURING SYSTEM SIMULATION.
5 = ASSIGN ELECTRODES TO RECEIVE DATA FROM OUTPUT FILES
6 = SAMPLE SPEECH OR OTHER DATA WITH THE A/D CONVERTER
7 = TEST A PATIENT BY SENDING DATA OUT OVER ASSIGNED ELECTRODE CHANNELS

ENTER > 3
ENTER DESIGN NUMBER (1-9999): 2
NOW READING INITIALIZATION DATA AND OPENING FILES
PROCESS ENTIRE INPUT FILE OR ONLY A PORTION OF IT? (E = 0; P = 1): 0
NOW SIMULATING NETWORK
SAMPLE NUMBER = 1024
SAMPLE NUMBER = 2048
SAMPLE NUMBER = 3072
SAMPLE NUMBER = 4096
SAMPLE NUMBER = 5120
SAMPLE NUMBER = 6144
SAMPLE NUMBER = 7168
NUMBER OF PAGES USED IN EXTENDED MEMORY = 14
TIME TO SIMULATE NETWORK: ETIME = 15 SECONDS
NOW CLOSING FILES

PAUSE _

```

Fig. II.2.22. Complete display of the feedback provided by program EXECUTE for the example system. The display indicates that the total number of 1024-word pages used in extended memory for the simulation was 14, and that the total time required for processing 7168 input samples was 15 seconds.

```

SELECT THE NEXT TASK FROM THE FOLLOWING OPTIONS:
1 = DESIGN A NEW SYSTEM
2 = MODIFY AN EXISTING SYSTEM
3 = EXECUTE SIMULATION OF AN EXISTING SYSTEM
4 = DISPLAY OUTPUTS WRITTEN TO THE DISK DURING SYSTEM SIMULATION
5 = ASSIGN ELECTRODES TO RECEIVE DATA FROM OUTPUT FILES
6 = SAMPLE SPEECH OR OTHER DATA WITH THE A/D CONVERTER
7 = TEST A PATIENT BY SENDING DATA OUT OVER ASSIGNED ELECTRODE CHANNELS

ENTER > 4_

```

Fig. II.2.23. Menu presented by CPEXEC. The option entered at the bottom indicates that the next program to be called is SHOWNTTELL (for the display of outputs written to the disk during system simulation).

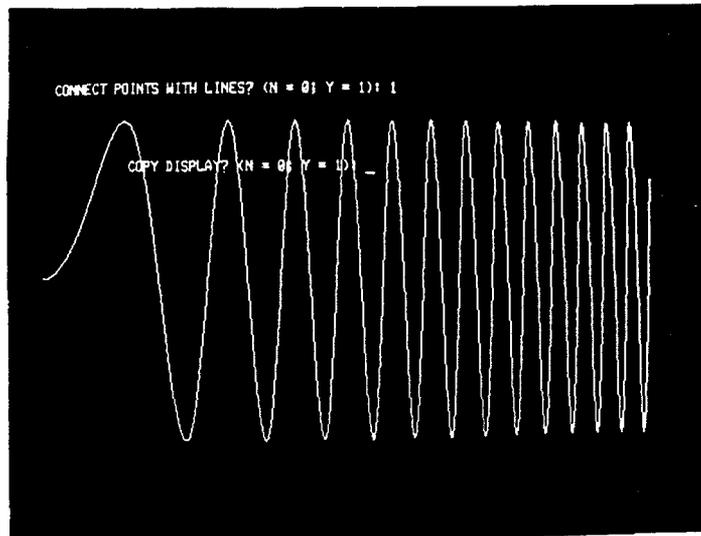


Fig. II.2.24. Display of the output of block # 1 in the example network of Fig. II.2.3.

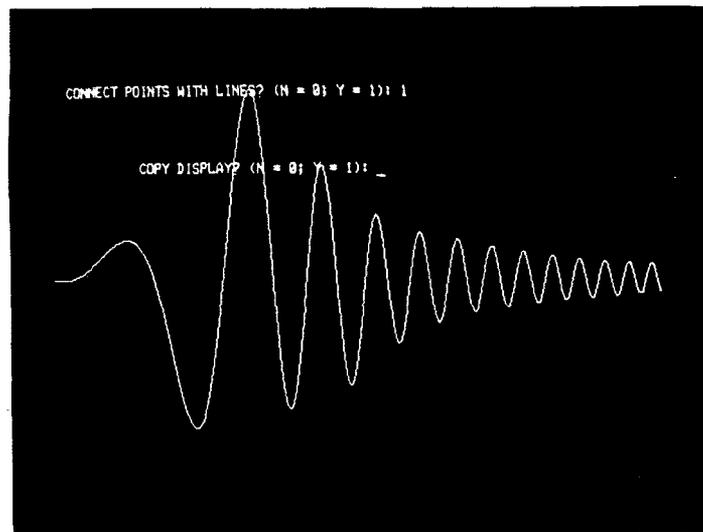


Fig. II.2.25. Display of the output of block # 2 in the example network of Fig. II.2.3.

encompass the break frequencies of the bandpass filters in the example network.

Only the initial 512 data points of TUNA are shown in Fig. II.2.24. SHOWTELL also has facilities for displaying other portions of the file;

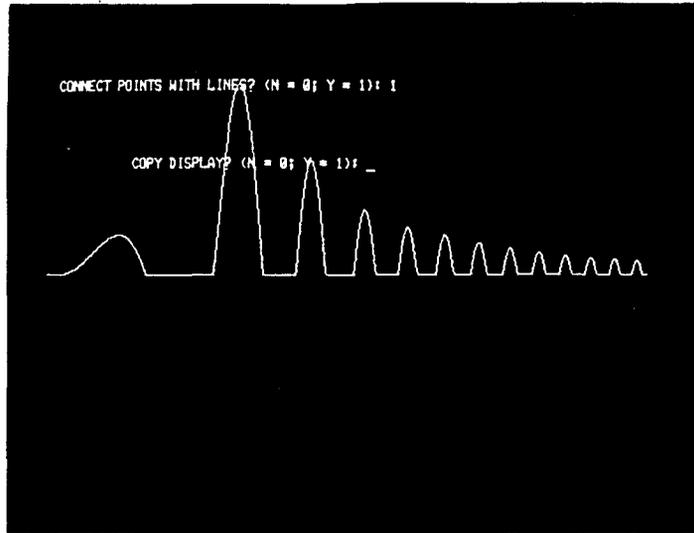


Fig. II.2.26. Display of the output of block # 6 in the example network of Fig. II.2.3.

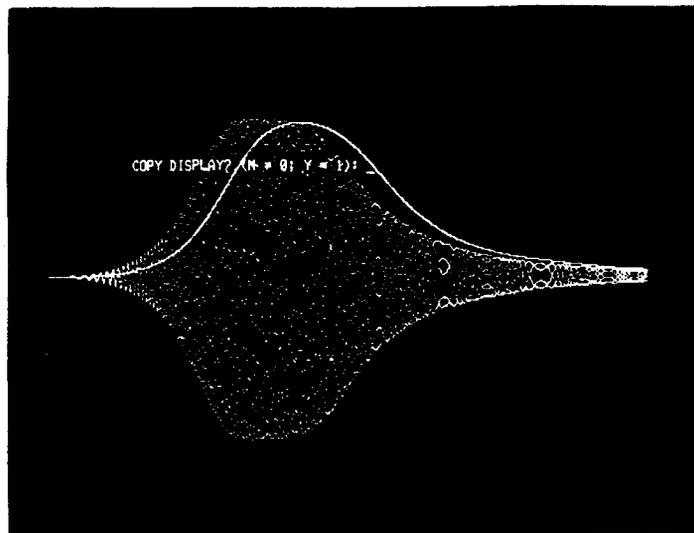


Fig. II.2.27. Display of the outputs of blocks 5 and 13 in the example network of Fig. II.2.3.

manipulating the number of points displayed; manipulating the y-axis scale factor of the display; interpolating or downsampling points in the display; and presenting data in multitrace and/or multioutput displays. Also, as

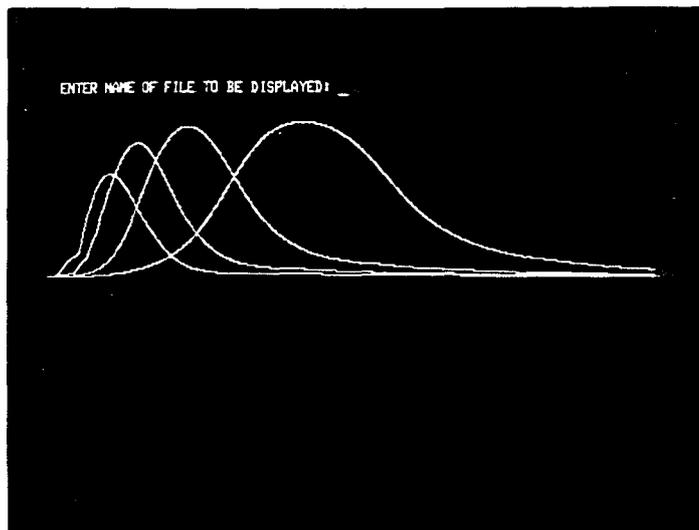


Fig. II.2.28. Display of the outputs of blocks 10 through 13 in the example network of Fig. II.2.3.

indicated in the initial description of SHOWNTTELL on p. 17, data held in files can be presented as acoustic signals using the D/A converter.

The outputs of key signal-processing blocks in the example network are displayed in Figs. II.2.25 through II.2.28. Fig. II.2.25 shows the output of block # 2, a fourth-order Butterworth bandpass filter with break frequencies at 100 and 400 Hz. As expected, the response of the filter is greatest when the instantaneous frequency of the input sweep falls within the filter passband.

The remaining displays are also consistent with the functions of the blocks in the network. For example, Fig. II.2.26 shows the output of block # 6, which is the rectified waveform of the output shown above for block # 2. Another example is presented in Fig. II.2.27. This display shows the first 2560 points of the outputs of blocks 5 and 13, where the smooth curve corresponds to the output of block 13. Clearly, the output of block 13 represents well the time course of rms energy present in the band from 1600 to 3000 Hz. The delay of output 13 relative to output 5 is a consequence of the sluggish response time (for this range of input frequencies) of the 30-Hz lowpass filter in block 13. Finally, Fig. II.2.28 is a composite of the outputs of all four channels in the example network. The trace with the earliest peak is the output of block # 10; the trace with the next earliest

peak is the output of block # 11; the trace with the third earliest peak is the output of block # 12; and the trace with the latest peak is the output of block # 13, as previously shown in Fig. II.2.27. This progression of responses reflects the time course of frequency changes in the input sweep.

Before leaving this subsection on the computer-based simulator of speech processors, we want to note the important point that use of this system will allow us to make valid comparisons between many different approaches to processor design in tests with individual implant subjects. That is, we can simulate every extant speech processor for auditory prostheses (as described in the open literature) and we can simulate most of the processors described in sections III.B and C of this proposal (the remaining ones can be simulated after we add in special signal-processing modules particular to those strategies). Within single testing sessions we can not only randomize the presentation of processed speech tokens, but also can randomize within blocks several separate strategies used to produce the processed tokens. In this way we can control for learning effects, for "biasing" effects (in terms of intelligibility testing) produced by patient preference for certain strategies, and for lability of psychophysical and speech-recognition judgments from one testing session to the next. Also, and perhaps more importantly, tests of many strategies with individual implant subjects will provide controls for 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), the type of electrode array used (As indicated in section II.B of this proposal, there are huge differences in the field patterns produced by different electrode arrays.), and apposition of individual monopolar or bipolar-pair electrodes to excitable tissue. These differences among subjects, along with differences in testing procedures among laboratories, have made ~~previous~~ ^{previous} comparisons of processing strategies a very difficult and largely unrewarding exercise.

2. Software for support of RTI Patient Stimulator

In designing the software handler that operates the RTI Patient Stimulator three clear priorities were observed:

1. to make available to a compiled FORTRAN program, running under Data General's Advanced Operating System (AOS), the full bandwidth capability of the Patient Stimulator, even for eight independent channels and a complex stimulus lasting several seconds;
2. to provide ^{an} ~~and~~ array of convenient software features that simplify both automatic and special-purpose coding of various types of stimuli, while preserving access to the full flexibility of the Patient Stimulator hardware;
3. to minimize the size of stimulus code files.

The device handler we have developed for the Patient Stimulator is called TUBE. It is a self-contained program for execution by a digital control unit (DCU) operating under the control of a Data General ^{Eclipse} ~~Elipse~~ computer. TUBE reads code from buffers in the main Eclipse memory and, as often as each 50 microseconds, interprets a control word imbedded in that code and outputs an appropriate command string to the RTI Patient Stimulator. In keeping with the priorities listed above, TUBE is designed to output as quickly as possible the number of stimulator commands specified in the current control word. TUBE then will fulfill any "homework assignment" made by the same control word (such as resetting TUBE's buffer pointer, requesting buffer service from a task running on the Eclipse itself, reading the Patient Stimulator's ADC registers, or halting the DCU). Alternatively, the control word may instruct TUBE to spend a prescribed length of time in a "coasting" mode--supplying the Patient Stimulator with appropriate continuation messages but not reading new control words.

TUBE is one of several device handlers that may be executed by the DCU. Others control analog-to-digital and digital-to-analog converters, for instance. Utility subroutines are available to load these handlers into the

DCU's dedicated memory and establish communications between them and the AOS tasks that will be using them. DCUSOLO is normally used to load TUBE, dedicating the full speed of the DCU to that single task.

When done in the context of Block-Diagram Compiler output, the whole process of converting a stimulus waveform to stimulator code and making it available to TUBE is automated and completely transparent to the user. In the discussion that follows, however, we will illustrate the process by referring to utilities that accomplish these tasks in stages, for program development and testing purposes.

BUILD is a FORTRAN program that defines a mnemonic language for creating special purpose stimulator code files directly--roughly the equivalent of an "assembly language" for the Patient Stimulator. This allows (and requires) the user to specify each control word and stimulator command of such a file.

Another program, CODER, generates stimulator code files that conform to a standard format also used by the block diagram compiler. The user supplies CODER with a digitized waveform file for each channel to be driven. The program then verifies that the files are compatible and creates a single optimized stimulator code file. In addition, the user may specify an initial configuration for the Patient Stimulator or indicate that any of a number of optional stored configurations may be used with this stimulator code. Such configurations (any number of optional ones may be generated using a program called CONFIG) include specification of stimulator clock rate, relay settings, ground connections, and ADC control words; verification that the correct hardware configuration plug is installed; and optional specification of a custom sequence of initialization commands to the Patient Stimulator.

POUR is a FORTRAN main program (compiling to an AOS task) that (1) uses DCUSOLO to install TUBE in the DCU and establish communications between POUR and TUBE, (2) sets up a windowed memory system for double buffering, (3) loads the beginning of a user-specified standard-format stimulator code file (one produced by CODER, for instance) into the buffers, (4) determines whether this file contains its own configuration back and, if not, requests the name of a CONFIG-produced file from the user and loads it, (5) signals TUBE to begin execution, and (6) services the buffers and terminates as requested by TUBE.

3. Software for support of basic psychophysical studies and speech testing

Speech testing is achieved by presenting speech data files that have been prepared by the block-diagram compiler. These speech files are the synthesized output from a simulated speech processor. Inputs to the processor are speech tokens from the "MAC", "miniMAC", Klatt synthesizer, and/or confusion matrices. All speech data files are computed offline and are assembled into disk files for rapid access during testing. With speech data files available on disk at test time, full randomizations of test token presentation are possible. In preparing the speech data files the final step is to process the files using a specialized compiler called CODER (see section II.C.2).

For most psychophysical studies the necessary stimulator code buffers must be generated by the very program coordinating the test sequence, rather than being read from a file prepared in advance. The design of TUBE (our Patient Stimulator control program, see section II.C.2 immediately above) makes such "real-time" synthesis easy to achieve. To illustrate this, we shall list some example psychophysical test stimuli, indicating how TUBE's features facilitate their rapid production.

Tests that utilize rectangular pulses in silence--hearing threshold as a function of either pulse amplitude or duration, loudness comparisons varying pulse amplitude or duration, etc.--are particularly easy to code. Varying the amplitude of a pulse of any length merely requires the altering of a single word in a buffer. (One buffer contains instructions for the current stimulus pulse(s) and is executed once by TUBE whenever the stimulus is to be presented. At other times TUBE repeatedly executes another buffer that generates only silence.) Altering pulse duration only involves moving a termination command from one location within the buffer to another. Using biphasic pulses merely doubles these modest computational loads

To produce a continuous noise stimulus we prepare three buffers: one with a noise onset (cosine bell envelope, for instance), a second with continuous noise, and a third in which noise is smoothly terminated. Then a noise stimulus of arbitrary length (as required, say, for an extinction test) easily can be provided by having TUBE execute buffer one once, buffer two repeatedly for as long as necessary, and, finally, buffer three. Noise pulses less than one buffer in duration can be calculated quickly by imposing an envelope on buffer two. Similarly, rectangular pulses can be

superimposed on continuous noise by adding a constant value to the appropriate subset of locations in a copy of buffer two and executing the copy once, preceded and followed by repeated execution of buffer two itself. The amount of computing necessary between stimuli does increase significantly in the case, say, of frequency bursts superimposed on noise, but the buffer switching capabilities of TUBE still minimize the number of locations that must be recomputed.

D. Patient Tests

Several experiments were performed at UCSF in mid-March, 1985, with patient EHT, ~~UCSF~~, who is fitted with the four-channel, transcutaneous transmission system designed by the UCSF team. This patient had been the subject of intensive studies in the previous year using both the transcutaneous and percutaneous systems for transmission of stimuli to his implanted electrode array. The main objectives of our studies with EHT were the following:

1. Obtain basic psychophysical measures of EHT's performance with the transcutaneous system, including measures on all channels of thresholds to pulses of various waveforms and duration^s, measures of temporal discrimination on three channels, measures of the loudnesses of pulses of various amplitudes and durations for two channels, measures of loudness matches for suprathreshold pulses of various waveforms and durations for two channels, measures of "pitch" and "sharpness" comparisons for suprathreshold pulses of various waveforms and durations for three channels, measures on all channels of thresholds to bursts of filtered noise (filter break frequencies were 2.0 and 6.0 kHz, with 4th order skirts beyond each break frequency), measures of the time for "decay" or "extinction" of initial percepts when the filtered noise was presented continuously, and measures of loudness difference limens (DLs) for pulses delivered alone and for pulses superposed on a continuous background of "extinguished" bandpass noise;
2. Obtain measures on the repeatability of measurements made with the transcutaneous system, including measures of the effects on apparent pulse thresholds of manipulations in the positioning of the antenna array and in the connections between rf modulators and "unused" coils in the antenna array;
3. Compare psychophysical measures of EHT's performance with the transcutaneous system to previous measures of his performance with the percutaneous system, and have available a "baseline" of data for the transcutaneous system for comparisons with future

- implant subjects;
4. Evaluate certain hypotheses that relate to the design of "stimulus primitives," using some of the basic psychophysical measures listed above;
 5. Simulate the present UCSF speech processor with the block-diagram compiler to confirm that speech-testing results obtained with the block-diagram compiler and hardware interface are essentially identical to the results obtained with the analog processor; and
 6. Confirm that all hardware and software components of the RTI testing facility work according to design and are "ready to go" for the next implant patient.

All of these objectives were met with the exception of objective 5. This objective was not realized because we had only eight, 5-hour days for completing the tests ~~with~~^{with} EHT and we simply ran out of time. However, we did meet the other objectives, including confirmation that all hardware and software components (with the exception of the block-diagram compiler simulation) of the RTI testing facility at UCSF work according to design and are ready for the next implant patient, to be intensively studied by ^{the} UCSF/RTI ^{team} this June and July. To conserve time during testing sessions with this next patient, we are automating many of the psychophysical procedures used to obtain the data indicated in point 1 above. In addition, we are in the process of making side-by-side comparisons of output^s and intermediate waveforms produced with the present analog UCSF speech processor and with the simulation of this processor using the block-diagram compiler system. So far, no discrepancies (other than very minor differences that would be expected between an analog system and a digital simulation of it) have been discovered between waveforms produced by the two processors, and we expect that the block-diagram compiler system will also be "ready to go" for this next patient.

For reviewers of this proposal interested in the details of our studies with EHT, we note that a complete description of the experiments and results listed above will be presented in our next quarterly report. Selected results are also presented in section III.A of this proposal, in regard to

the design and evaluation of "stimulus primitives."

In concluding this subsection on patient testing, we would like to acknowledge the many contributions of Dr. Mark White of UCSF to the conduct and design of these experiments.

E. Development of Portable, Real-Time Hardware

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 (1) 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; (2) 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; (3) 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 (for more on this see sections III.B and III.C of this proposal); (4) provide a working hardware system for implementing other promising strategies in a portable unit; and (5) 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, see sections III.B and III.D).

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. II.E.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 throughput" 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

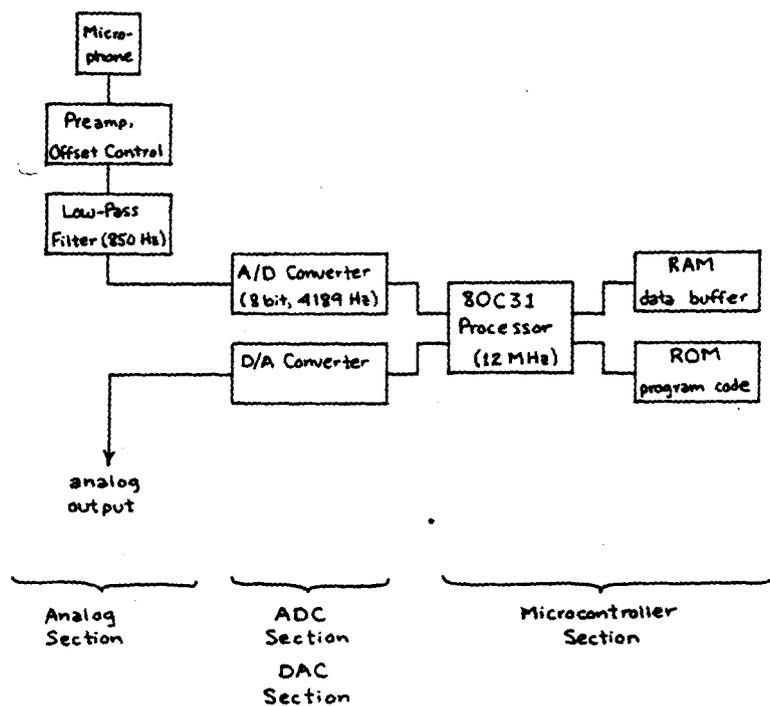


fig. II.E.1. Block diagram of the 80C31-based processor. See text for details.

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. [This estimate neglects, of course, the additional power required to drive prosthesis electrode(s).]

Also compatible with the objective of a portable unit is the small size of the instrument. A photograph of the prototype is presented in Fig. II.E.2. Even with the low density construction shown, the entire processor easily fits on a 12 x 9 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 our implementation of this algorithm the AMDF output is further processed for median smoothing, detection of erroneous indications of F_0 , and detection and signalling of intervals that contain unvoiced speech sounds. Informal tests with inputs of sinusoids, noise and speech material indicate that the processor functions according to its design. Formal tests are now underway with synthesized speech tokens and with digitized natural speech whose F_0 contours and voice/unvoice boundaries are fully known.

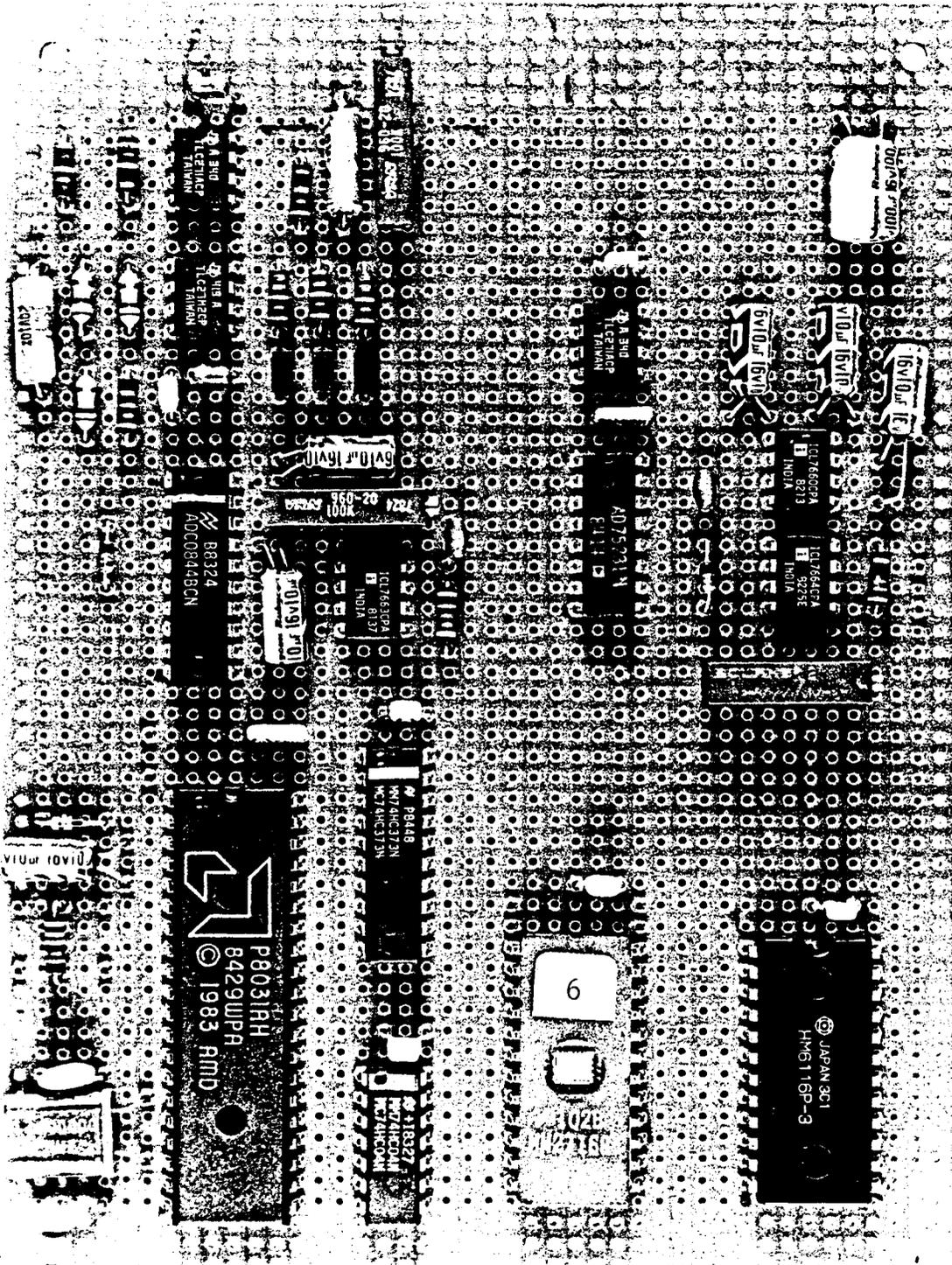


Fig. II.E.2. Photograph of the hardware prototype of a speech processor appropriate for use in a single-channel auditory prosthesis.

F. Collaborations

Our main collaboration for our present project is with the cochlear-implant team at the University of California at San Francisco (UCSF). We are pleased to acknowledge here the many contributions our coworkers at UCSF have made to our joint effort. All members of the UCSF team have generously shared with us hard-won knowledge gained through years of experience in the development of auditory prostheses, and have inspired us with the clear excellence of their work.

As indicated elsewhere in this proposal, we are working with the UCSF team to evaluate speech coding strategies for patients implanted with the UCSF multichannel electrode array. The RTI Patient Stimulator (section II.A), "Block-Diagram Compiler" (section II.C.1), and software for the support of basic psychophysical studies (section II.C.2,3), have all been installed and tested at UCSF. Software for support of tests to measure speech understanding (section II.C.3) is still under active development. As mentioned above (section II.D), we have tested one patient in San Francisco, and confirmed that all hardware and most of the software just listed works according to design. The next patient for the "experimental series" was implanted at UCSF on May 8, 1985. This patient will be intensively studied by the UCSF and RTI teams using the block-diagram compiler and other tools we have jointly developed. We expect to begin these studies in late May or early June, 1985.

Another collaboration we have, with Storz Instrument Company of St. Louis, is related to our collaboration with UCSF in that Storz is the manufacturer of the UCSF electrode array and present version of the speech processor. This speech processor is given to patients participating in the experimental series after studies with the percutaneous cable have been completed (usually within 3 or 4 months postimplant). Therefore, these subjects receive a state-of-the-art processor at the end of the testing period. Storz has agreed to work with us in implementing major advances we (RTI and UCSF) may make in processor design into commercially-available devices.

An additional aspect of our collaboration with Storz is that Storz has funded RTI to conduct a modest feasibility study "to evaluate the efficacy of single-channel coding strategies for extra-cochlear auditory prostheses." This funding provides separate support for evaluation of these particular

strategies, as required in the present RFP. (Support for evaluation of stimulus primitives and multichannel coding strategies comes from other sources, see sections III.D and II.2.C.)

The final collaboration we have in our present project is with the Duke University Medical Center (DUMC). There, we have helped to establish the "Center for the Severely Hearing Impaired." A major part of the Center's work will be to implant suitable patients with the UCSF prosthesis, as supplied by Storz Instrument Company, and to conduct tests with these patients that follow the general plan of our tests with patients at UCSF. All patients implanted at Duke will be fitted with the percutaneous cable system, allowing a broad range of studies with our block-diagram compiler and other tools. In addition to percutaneous access to the electrodes in the implanted scala-tympani array, an "extra" cable will be routed through the UCSF disconnect pad to allow placement of an extracochlear electrode near the round-window niche. This arrangement will allow us to compare, in single subjects: single-channel coding strategies as applied with an extracochlear electrode, single-channel coding strategies as applied with an intrascalar electrode or electrodes, and multichannel coding strategies as applied with the intrascalar electrode array.*

To accelerate the schedule for testing patients at DUMC, RTI's Neuroscience Program Office and Duke's Department of Surgery have jointly funded the construction of a cochlear-implant laboratory at Duke. This laboratory is functionally identical to the laboratory we have helped to establish at UCSF. It contains an Eclipse S/140 computer system (on loan to Duke from RTI) and a hardware interface for communication between the computer and implanted electrodes. The laboratory is in the final stages of construction and will be ready for tests with the first implant patient at Duke. This first implant is expected to be in mid-June, 1985.

*Of course, the presence of the intrascalar array and silastic carrier is likely to distort somewhat the fields produced in the ear by the round-window electrode. We are examining the extent to which such an effect may complicate the interpretation of results obtained from comparisons of speech understanding with different processors and electrode configurations. Preliminary results from our field-mapping model (section II.B.1) suggest that the effects of a passive intrascalar implant on the electric fields produced by the round-window electrode will be very small.

III. Plan of the Proposed Effort

In this section we will respond to the specific requirements of RFP No. NIH-NINDCS-85-09, as presented in article C.2 of the RFP, "Statement of Work."

First, we will describe our work to date on the design of "stimulus primitives," single-channel coding strategies and multichannel coding strategies. As mentioned in the INTRODUCTION of this proposal, the overall goal of our work on stimulus primitives is to improve the representation of fundamental dimensions of auditory stimuli such as intensity and frequency. Our specific objectives for this work are the following:

1. Improve the temporal and spatial resolution of neural excitation via cochlear implants;
2. Extend the dynamic ranges of intensity and frequency coding;
3. Preserve constant percepts of pitch and timbre while changing percepts of loudness;
4. Increase the salience of pitch percepts;
5. Improve our present understanding of the factors and mechanisms that control the subjective sensation of pitch.

These objectives are essentially identical to those indicated in section I.A of the RFP work statement. Our plan for implementing studies to address the objectives just listed is presented in sections III.A and III.D of this proposal.

Hypotheses of potentially-useful strategies for coding and presentation of speech information with auditory prostheses are described in section III.B for single-channel devices and in section III.C for multichannel devices. The main objectives of our work on single-channel coding strategies are to (1) develop single-channel, extracochlear auditory prostheses that can be safely and efficaciously applied in infants and young children; (2) improve methods for coding speech signals that reflect the

excitation of the vocal tract; and (3) provide a "building block" for multichannel speech processors in which signals representing the excitation of the vocal tract are coded separately from signals representing the "short-time" configuration of the vocal tract (e.g., the basic Australian strategy and variants, see section III.C).

Realization of some or all of the objectives listed above for stimulus primitives and single-channel coding strategies would, of course, improve our chances for meeting the primary objective of this proposed project, which is to define the classes and parameters of processor design that will allow full recognition of speech for recipients of multichannel implants. A detailed outline of multichannel coding strategies that have promise for meeting this objective (in terms of our present knowledge) is presented in section III.C. Also presented in section III.C are (1) proposed procedures for characterizing the "electrical-to-neural transformer" in patients implanted with the UCSF multichannel electrode array; (2) tests to evaluate degradations in performance of multichannel processors when interfering signals are added to the speech input; and (3) strategies to improve the performance of various multichannel processors for "real-world" acoustic environments with significant reverberation and noise.

The sections on design of single-channel and multichannel coding strategies address point I.B.3 of the RFP work statement, which is to "devise hypotheses of potentially feasible processing schemes based on presently known psychophysical data from auditory prosthesis implant patients." The strategies presented in these sections are representative of our present thinking; of course, we expect that knowledge obtained in our future work will lead to refinement of the present strategies and to development of new and better strategies.

The major tools required by the collaborating psychophysical teams to evaluate stimulus primitives and speech-coding strategies are outlined in points B, B.1, B.2, B.3, and B.3.a-e of the RFP work statement. Most of these tools have already been developed by us in our present contract. In particular, the tools we now have operational and in place in our laboratories in San Francisco and North Carolina include the following:

1. A hardware interface that provides a safe, high-bandwidth communications link between our Eclipse computers and implanted electrodes (see section II.A of this proposal);

2. A computer-based system that is capable of rapid and practical simulation of every extant processor for auditory prostheses (as described in the open literature) and most of the processors presented in sections III.B and C of this proposal (for a full description of the "block-diagram compiler," see section II.C.1);
3. Various computer programs for support of basic psychophysical studies and for support of studies in which measures of speech intelligibility and recognition are obtained (see sections II.C.2 and 3).

The proposed project, for a possible contract to follow the present one, would allow us to (1) add the necessary modules to the computer-based simulator so that all extant and suggested new processors can be tested in comparisons with single implant subjects; (2) refine the computer-based simulator to improve investigator interaction with the implemented models of processing strategies; (3) refine, if necessary, the hardware we have designed but not tested for obtaining intracochlear evoked potentials; (4) write software for the generation of signals to evaluate stimulus primitives, as will be further described in section III.A of this proposal; and (5) complete our work on the many computer programs that will be required to support the efforts of the collaborating psychophysical teams.

An additional requirement of the RFP, related to the evaluation of stimulus primitives and speech-coding strategies, is that "the offeror must have separate financial means of supporting the evaluation of the speech processing schemes developed under this contract even though tentative plans for this evaluation are required in the proposal." Support for the evaluation studies is outlined in sections II.F, "Collaborations," and IV.C, "Collaborative Arrangements," of this proposal. A brief description of tentative plans we will share with the collaborating psychophysical teams for the evaluation studies is presented in section III.D, "Experimental Plan."

Next, sections I.C and I.D of the RFP work statements specify the delivery of "wearable speech processors based on results obtained with the computer-based simulated designs." These processors are to be (1) "designed

for specific patients with single or multielectrode auditory prostheses; (2) human engineered with respect to weight, durability, and panel component selection and placement; (3) take advantage of the implanted electrode configurations; and (4) operate in real time." In addition, the processors must be suitable for use in studies to evaluate possible long-term effects of learning with a given patient and processing strategy. Finally, the RFP work statement requires that at least two of these wearable processors be delivered to the Project Officer within three years after the beginning of the contract.

Our response to these requirements is presented in section III.E of this proposal; "Further Development of Portable, Real-Time Hardware." As indicated in section II.E, we have direct experience in building wearable speech processors for auditory prostheses and, as indicated in sections III.E and IV.B, we have extensive experience in the design and fabrication of microprocessor-based, battery-powered instruments for real-time analysis of speech and other signals. In the "present-scope" proposal we have outlined a plan for development of portable speech processors that use components of hardware systems we have already designed and debugged. In this "expanded-scope" proposal we will also outline plans for full evaluation of alternative approaches. If one or more of these alternative approaches emerges as superior to the already-good approaches we have developed here at RTI, then the superior approach or approaches will be implemented in portable hardware devices.

Finally, in section III.F, we will offer our views on the prospects and likely outcomes of the work outlined in this proposal.

A. Design of Stimulus Primitives

The main purpose of our studies on "stimulus primitives" is to extend the present limits of psychophysical performance exhibited by implant patients. These limits have been determined in many studies (see, e.g., Bilger, 1977 and 1983; Eddington et al., 1978; Pfingst et al., 1983, 1984 and 1985; Shannon, 1983a; Simmons, 1966; Simmons et al., 1979b; Tong and Clark, 1985), and have been well summarized by Muller in recent publications (Muller, 1981 and 1983). Briefly, complex percepts of sounds, that can be roughly ranked along a scale of pitch or timbre, are elicited according to periodicity of stimulation and electrode position. For a given electrode (either monopolar or a single bipolar pair) and intensity of stimulation, the percept of pitch follows the frequency of sinusoids, or the rate at which pulses are delivered, up to about 300 Hz (the range is 150-1000 Hz, see Muller, 1983). The difference limens (DLs) for this repetition or "volley" pitch are often 5% or less for frequencies below 300 Hz, but DLs rise rapidly as frequency is increased much beyond this limit of "pitch saturation." Also, results of scaling and matching experiments indicate that, while pitch corresponds to rate for stimulus frequencies up to about 300 Hz, it either accelerates to very high values (Eddington et al., 1978; Simmons, 1979b) or does not increase (Tong et al., 1979) thereafter. Useful encoding of speech parameters along the dimension of rate or volley pitch percepts is therefore probably limited to frequencies below 300 Hz.

When the frequency and intensity of electrical stimuli are held constant, but the site at which stimuli are delivered is varied, distinct tonal sensations (e.g., along a scale of "sharp" to "dull" in tone) are evoked that can be ranked according to the electrode's position along the cochlear partition (Chouard and MacLeod, 1976; Eddington et al., 1978 and 1980; Hochmair et al., 1979; House and Edgerton, 1982; House and Urban, 1973; Mladejovsky et al., 1975; Tong et al., 1982 and 1983; Tong and Clark, 1985) or location within the auditory nerve (Simmons, 1966; Simmons et al., 1979a and b). To the extent that two mechanisms of pitch perception are involved in the complex sensations evoked by electrical stimuli (i.e., those corresponding to the rate or frequency of stimulation and those corresponding to the place at which stimuli are delivered), several investigators have proposed strategies in which separate elements of speech are coded along the dimensions of rate and place (Tong et al., 1982 and

1983; Tong and Clark, 1985; also see Appendix 1).

A further complexity of "pitch" perception is that apparent pitch is a function of stimulus intensity for both modiolar (Simmons, 1966; Simmons et al., 1979b) and scala-tympani prostheses (Pfungst et al., 1985; Shannon, 1983a). That is, for a given electrode and stimulus waveform, increases in stimulus intensity are usually accompanied by increases in loudness and pitch. Furthermore, alterations in the stimulus waveform can have large effects on the loudness, pitch and timbre of electrically-evoked percepts. Examples include the findings that (1) the perceived pitch of pulses delivered to single electrodes in a scala-tympani array can increase when pulse duration is decreased (Eddington et al., 1978); (2) distinct changes in the timbre or quality of the evoked sensations (along with changes in loudness) are produced when the durations of constant-charge pulses are manipulated over the range of 0.1 to 1.0 msec (Shannon, 1983a); and (3) pitch, threshold and loudness can all change in complex ways when the phase relationships between components in a seven-harmonic signal are manipulated (Muller, 1983). Finally, there is evidence that some high-frequency information in various waveforms is perceived, but not necessarily as changes in pitch or timbre. This evidence includes (1) the observations by the Hochmairs that speech intelligibility declines when energy between 900 and 4000 Hz is removed from stimulus waveforms derived by their single-channel speech processor (Hochmair and Hochmair-Desoyer, 1985); (2) the demonstration by Mark White that changes in the frequency of the first formant can be perceived by implant patients with a single channel of stimulation (White, 1983); and (3) the demonstration by Dobie and Dillier that triangular and trapezoidal waveforms can be discriminated from square waves even when the differences in rise times are as low as 80 microseconds (Dobie and Dillier, 1984).

Another major dimension along which electrical stimuli are perceived by implant patients is loudness. We have already mentioned some of the interactions between pitch and loudness for changes in stimulus intensity and manipulations in the stimulus waveform. Now we will further note the observations that (1) the dynamic range of electrical stimulation is also a function of waveform and frequency, but is generally between 8 and 20 dB; (2) this dynamic range is much narrower than the wide dynamic range of normal hearing; (3) however, the intensity DLs for electrically-evoked hearing are generally a little better than intensity DLs in normal hearing

(typical DLs for electrically-evoked hearing are between 0.5 and 1.5 dB for stimuli in the middle of the dynamic range, see Muller, 1983 and Pfingst et al., 1983); and (4) despite the somewhat better DLs, the number of discriminable intensity steps in electrically-evoked hearing is lower (sometimes much lower) than in normal hearing.

As indicated in the introduction to section III of this proposal, our specific objectives for work to develop and evaluate stimulus primitives are the following:

1. Improve the temporal and spatial resolution of neural excitation via cochlear implants;
2. Extend the dynamic ranges of intensity and frequency coding;
3. Preserve constant percepts of pitch and timbre while changing percepts of loudness;
4. Increase the salience of pitch percepts;
5. Improve our present understanding of the factors and mechanisms that control the subjective sensation of pitch.

Generally, our objective is to identify and then mimic the patterns of neural discharge that allow the normal auditory system to (1) perceive the intensities and frequencies of sounds along largely independent psychological dimensions; (2) resolve differences in these stimulus attributes along a finer grain within the dynamic range than is possible with electrically-evoked hearing; (3) perceive sinusoids and other simple sounds as "pure" in pitch or tone; and (4) discriminate well small changes in the frequencies of sounds in the "speech range," from about 300 to 3000 Hz. As should be obvious from the previous discussion on implant psychophysics, performance with cochlear implants falls far short of normal

hearing. Particularly important limitations that constrain the coding of speech information are the following:

1. "pure tone" pitches are not elicited with cochlear implants;
2. changes in "sharpness-dullness" in the percepts evoked by different electrodes in a multielectrode array are much more subtle and less discriminable than might be expected from "place-pitch" theory;
3. "repetition-rate" pitch saturates at about 300 Hz for implant patients; and
4. percepts of loudness, pitch and timbre covary with changes in a single attribute of the stimulus, such as the complex interactions reviewed above for changes in stimulus intensity.

Our approach for addressing the general problem just outlined is to model the patterns of neural discharge produced in electrically-evoked hearing (see section II.B) and then to mimic aspects of normal discharge patterns with implant patients. As will be described in the remainder of this subsection, normal perception of pitch and loudness probably depends on one or more (probably more) of the following aspects of neural discharge in the auditory periphery: (1) coincidence of "place" and "volley" information; (2) full representation of volley information in stochastically-independent discharges of fibers innervating the same region of the cochlear partition; (3) coincidence of "phase-shear" (in the "terminal region" of basilar-membrane displacements), place and volley information; (4) coincidence of temporal inputs from widely-separated segments of the cochlear partition, that could code frequency information by cross correlation of these inputs at the cochlear nucleus or higher centers; (5) preservation of sharp edges in the excitation fields produced by stimulation with sinusoids and complex sounds. Our present strategies for mimicking many of these aspects of normal discharge patterns in electrically-evoked hearing are

- manipulation of latency profiles to code loudness;
- manipulation of latency profiles to code pitch, with the aim of representing (a) phase breaks in the terminal region and (b) spatial cross-correlation of basilar-membrane phases;
- unidirectional coding of loudness, with the aim of maintaining a "sharp edge" in the excitation field for a constant percept of pitch across loudness;
- noise "biasing," to increase the number of discriminable steps on the loudness scale, to represent periodicity pitch as well as rate pitch (i.e., to push the limit of "pitch saturation" on single channels beyond 300 Hz), and to increase the salience and "purity" of pitch percepts; and
- manipulation of stimulus waveforms to improve the spatial resolution of excitation.

In the following subsections we will first describe a simple field-neuron model and then describe in somewhat greater detail the strategies listed above. The field-neuron model illustrates principles of several of these strategies.

1. Simple model of neural excitation by intracochlear electrodes

To describe characteristics of the excitation fields produced by intracochlear electrical stimulation, we will present in this subsection a simple model of neural stimulation with electrodes in the scala tympani. As will be evident from comparisons of results obtained with this simple model and the more-realistic models of section II.B, the simple model should be viewed only as a heuristic to understand the principles of the coding strategies to be outlined. Our more-sophisticated models clearly indicate that all the described strategies can be implemented in auditory prostheses, but in ways that are more complex than can be fully described in a brief presentation. Therefore we will present a framework for brief description

here with the understanding that the interested reader can seek more detail in section II.B.

Useful models of electrical stimulation of the auditory nerve must, of course, couple a description of the electric fields produced by the electrode(s) with a description of neural responses to such fields. An extremely simple yet powerful approach to this modelling problem is to couple a mathematical description of the field patterns, ~~calculated~~^{measured} for various electrode configurations, with a mathematical description of the ~~measured~~^{measured} strength-duration curves of electrical stimulation, also^{measured} for various electrode configurations and stimulus waveforms (in addition to the "classic" measurement with monophasic pulses). The resulting model provides an approximate picture of neural response fields produced with single electrical pulses. Refractory effects are not contained in the description and therefore neural discharges that follow the "first pop" are not modeled. However, as we will show, this "first pop" model can provide useful insight into the patterns of neural responses evoked by intracochlear electrical stimulation.

Falloffs in the electric fields produced by intracochlear electrodes have been measured directly or indirectly by several investigators. An example of one such set of measurements is presented in Fig. III.A.1 for 12 well-positioned and 3 mispositioned bipolar electrodes in the scala tympani of the cat (Merzenich and White, 1977). The lines are least-squares fits to thresholds of neural responses vs. place (the "frequency" axis) or distance from the electrode pair, as derived by the technique of Merzenich and White. Each division along the x-axis corresponds to one octave, or approximately 3 mm of frequency representation along the basilar partition of the cat. The least-squares lines for the well-positioned electrodes are on the left and the lines for mispositioned (e.g., in the middle of the scala tympani instead of directly under the osseous spiral lamina) are on the right. The slopes of the lines for the well-positioned electrodes indicate that the excitation field falls off at the rate of 10 dB/mm. This is a relatively high rate of falloff for intracochlear electrodes and demonstrates the high spatial resolution of excitation afforded by the electrode placements used in the UCSF electrode array (for much more on this, see section II.B).

Comparison of the approximate field pattern produced around each electrode pair of the UCSF array with the field patterns produced by other arrangements of electrodes is shown in Fig. III.A.2. In this figure the

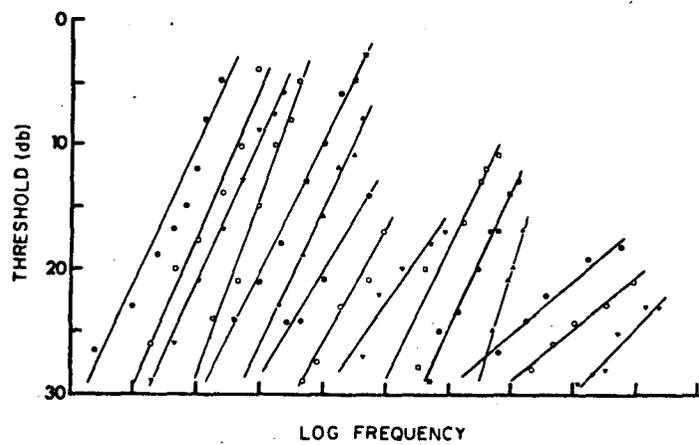
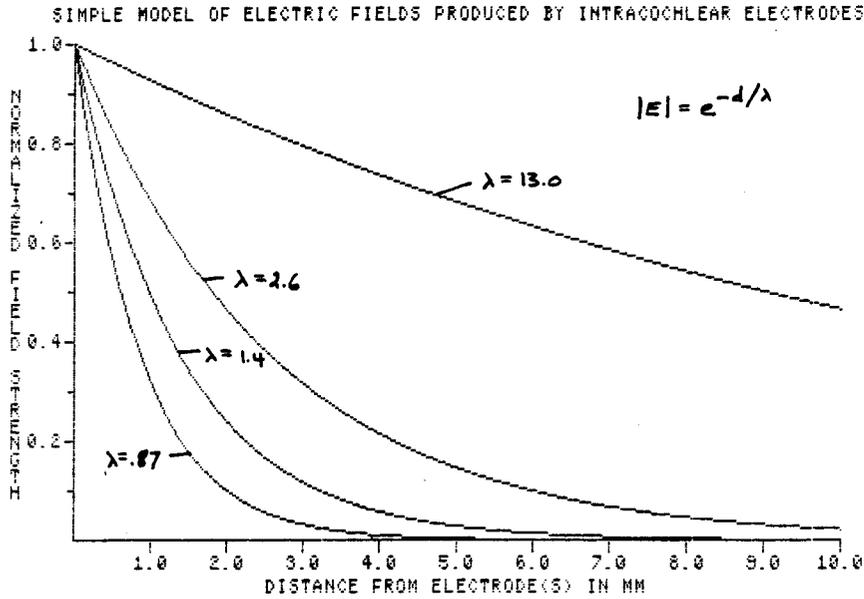


Fig. III.A.1. Thresholds of neural responses vs. place (the "frequency" axis) for intracochlear electrical stimulation, as derived by the technique of Merzenich and White (1977). See text for details.



lambda	remark
1.4	6 dB/mm dropoff, as found by v. Bekesy
2.0 - 4.0	4.3 - 2.2 dB/mm, as found for pseudo-bipolar stimulation by Black and Clark
0.87 - 2.8	10 - 3.3 dB/mm for stimulation of well-placed bipolar electrodes in the scala tympani, White and Merzenich, 1977
13.0	0.67 dB/mm, as found by Black and Clark for monopolar stimulation

Fig. III.A.2. Simple model of electric fields produced by intracochlear electrodes. Space constants for exponential falloff in the modeled fields range from .87 mm to 13.0 mm. As indicated at the bottom of the figure, these values correspond to those measured for various arrangements and types of electrodes in the scala tympani.

space constants for exponential falloffs are indicated for each trace in the upper panel and listed in the bottom table to match each space constant with the source of measurement data. As is obvious from the comparison, the fields produced by the UCSF electrodes are generally sharper than the fields produced by the "pseudo-bipolar" pairs of the Australian array (Black and Clark, 1980) and much sharper than the fields produced by monopolar electrodes.

The other component of useful models of electrical stimulation of the auditory nerve by intracochlear electrodes is a description of neural responses to the current fields produced by the electrodes. Strength-duration curves for neurons stimulated by constant-current pulses delivered to intracochlear electrodes have been measured by Loeb et al. (1983) and by van den Honert and Stypulkowski (1984). The data from Loeb et al. are presented in the top panel of Fig. III.A.3. These data indicate, for the electrode placements of the UCSF electrode array, that chronaxies (as measured for units within the anteroventral cochlear nucleus) range from 200 to 850 usec for monophasic pulses and that rheobase approximates 15 uA. The data of van den Honert and Stypulkowski indicate somewhat lower chronaxies and much higher rheobases for their experimental conditions (different placement of electrodes and monitoring of single units in the auditory nerve instead of the AVCN).

Model curves for the data of Loeb et al. (corresponding to the UCSF array) are presented in the bottom panel of Fig. III.A.3. Rheobase is 15 uA for all curves and chronaxies increase from 100 usec to 800 usec for the lower-left to the upper-right curves, respectively. A "typical" curve is next to the curve for the 800 usec chronaxie, and models thresholds of responses for neurons with a rheobase of 15 uA and a chronaxie of 400 usec.

Results obtained from combining the two model elements just described, of the field patterns and neural responses, are presented in Fig. III.A.4. The length constant for all traces is 1.4 mm, corresponding to moderately-well placed electrodes in the scala tympani, and rheobase and chronaxie are 15 uA and 400 usec, corresponding to typical data reported by Loeb et al. In the top panel the amplitudes of 500 usec pulses delivered to intracochlear electrodes are manipulated and in the bottom panel the durations of 200 uA pulses are manipulated. The neural response fields produced by these stimuli are indicated by the curves in each panel. The "latency" of response is the time after pulse onset that corresponds to the

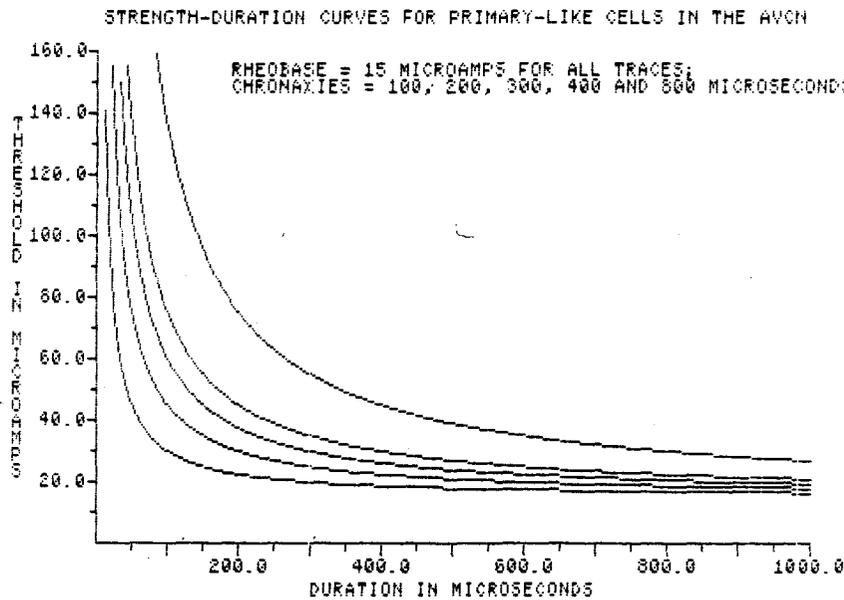
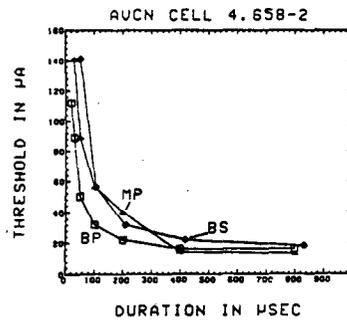
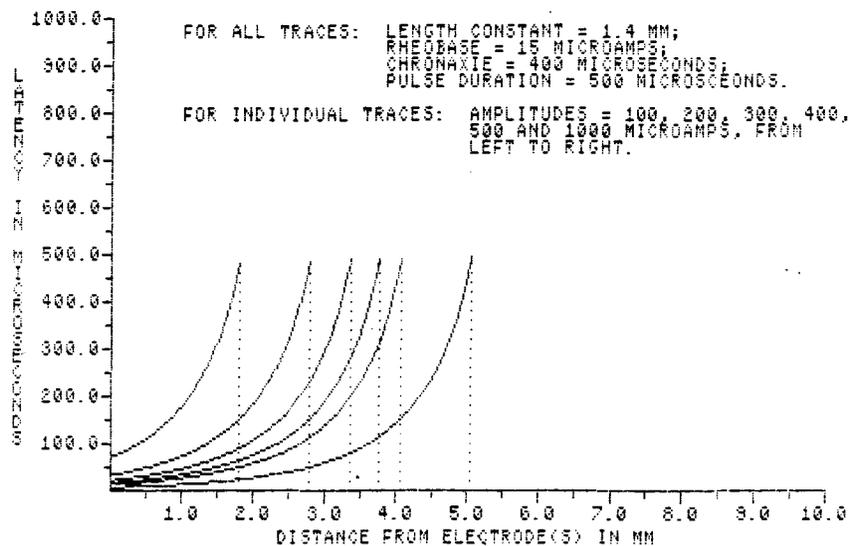


Fig. III.A.3. Top panel: Data from Loeb et al. (1983) on the strength-duration curves of cells in the anteroventral cochlear nucleus (AVCN) that responded to intracochlear electrical stimulation. Curve MP indicates thresholds of responses to monopolar, monophasic cathodal stimulation (single pulses at 8 pps); BP indicates thresholds to bipolar, monophasic cathodal stimulation; and BS indicates thresholds to bipolar, biphasic stimulation using a continuous sinusoidal waveform whose half-cycle duration is plotted on the x-axis. Bottom panel: model of strength-duration curves shown in the top panel.

RESPONSE FIELDS OF NEURONS EXCITED BY INTRACOCHLEAR ELECTRODES



RESPONSE FIELDS OF NEURONS EXCITED BY INTRACOCHLEAR ELECTRODES

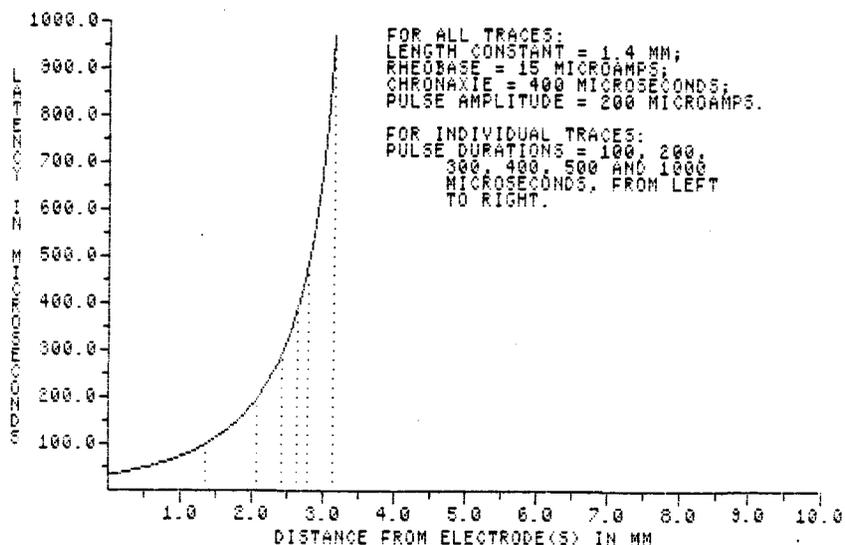


Fig. III.A.4. Response fields of neurons excited by intracochlear electrodes, as predicted from simple models of the falloffs in electric fields produced by intracochlear electrodes and the strength-duration curves of stimulated neurons. See text for details.

instant at which the threshold of the strength-duration curve is crossed. Actual latencies would be longer than those indicated because the threshold crossing only marks the initiation of spike discharge. However, the forms of the actual latency fields will approximate the forms of the "latency" fields shown in Fig. III.A.4.

As one might expect, a parabolic-like profile of latencies is produced by stimulation with monophasic, rectangular pulses. That is, as distance from the stimulating electrode (or electrode pair) increases, the electric field falls off and the neurons at these locations are stimulated further and further out along their strength-duration curves. Ultimately, the strength of the current field falls below threshold (for the duration of the pulse) and neurons at locations more distant than this point are not stimulated. This boundary is marked by the vertical dotted lines in Fig. III.A.4. Only one-half of the response field is shown in the figure; for this model a symmetrical pattern of responses ~~are~~^{is} present "to the left" of the electrode or electrode pair at location 0.

Stimulation with pulses of constant duration but of various amplitudes produces effects like those illustrated in the top panel of Fig. III.A.4. For low-amplitude pulses a relatively-small patch of neurons is excited and for high-amplitude pulses a broader patch of neurons is excited. However, the increase in the extent of the excitation field is not a linear function of pulse amplitude; indeed, one can see a clear "compression" of growth in the excitation field with increases in stimulus intensity.

Another potentially-significant effect of increases in stimulus intensity is that the shape of the latency profile is different for low-intensity and high-intensity pulses. Specifically, high-intensity pulses produce a relatively-large field of nearly-synchronous responses while low-intensity pulses do not. In the example shown in the top panel of Fig. III.A.4, 1000 uA pulses produce a highly-synchronous response field for neurons within a distance of 2.5 mm from the electrode. As we will describe in the following subsections, such differences in latency profiles could have important perceptual correlates.

Finally, stimulation with pulses of constant amplitude but of various durations also produces increase^s in the width of the excitation field as the amount of charge in the pulses is increased. Again, the increase in the width of the excitation field is a nonlinear function of charge (in this case, of pulse duration and in the case of the top panel, of pulse

amplitude) and the synchronicity of evoked neural activity changes as pulse duration changes. For short-duration pulses a relatively synchronous field is produced, and for long-duration pulses a long "tail" of asynchronous activity is "attached" to the region of synchronous discharges in the immediate vicinity of the electrode.

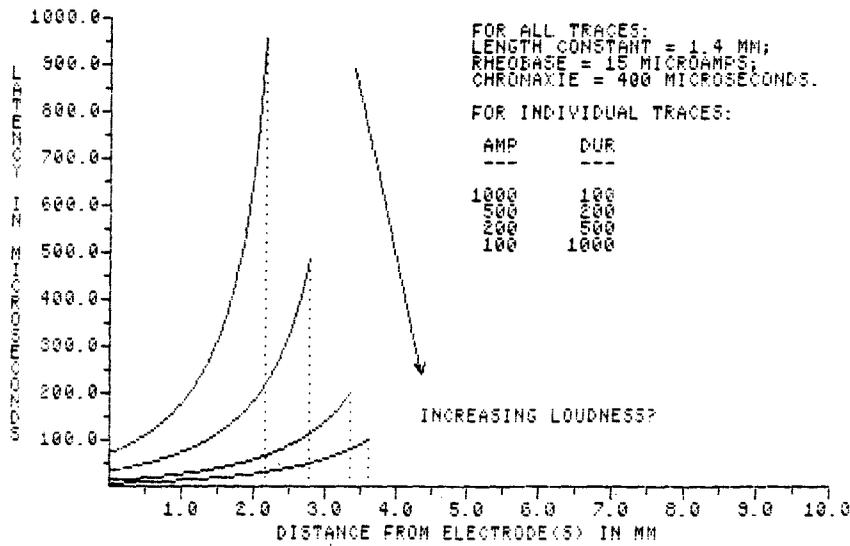
Because all these stimuli produce relatively-synchronous excitation fields in the immediate vicinity of the electrode, and because only a small field of excitation is likely to be required to elicit a threshold response for an implant patient, one might expect psychophysical thresholds to lie along lines of constant charge for intracochlear stimulation. This expectation is, in fact, borne out by the results of many studies (see, e.g., Muller, 1983; Shannon, 1983a). However, for various suprathreshold stimuli of constant charge the latency profiles and the extents of excitation can be quite different. In the next subsection we will present some possible consequences of these differences.

2. Manipulation of latency profiles to code loudness

The top of panel of Fig. III.A.5 shows the excitation fields predicted for constant-charge pulses of various durations and amplitudes. Decreases in duration (and increases in amplitude) produce broader, more synchronous response fields. One might therefore expect that, for suprathreshold stimuli, short-duration pulses would be louder than long-duration pulses of the same charge. In fact, as mentioned before, perceived loudness falls off rapidly as constant-charge pulses are increased in duration from 0.1 to 1.0 ms (Shannon, 1983a). In addition, these manipulations produce distinct changes in the quality of the perceived "sound" that are not the same as changes associated with simply decreasing the amplitude ^{of} ~~on~~ constant-duration pulses.

These model predictions and psychophysical findings suggest that either extent of the excitation field (or total number of neurons stimulated) or synchronicity of input or both contribute to the percept of loudness. If synchronicity of input contributes to loudness, then manipulations of the type illustrated in the bottom panel of Fig. III.A.5 would be useful for coding the intensities of sounds for auditory prostheses. That is, intensities could be coded, at least in part, by changing the synchronicity of discharge activity over a constant-width segment of the basilar partition. By keeping the "edges" of the excitation field at constant and

RESPONSE FIELDS OF NEURONS EXCITED BY INTRACOCCHLEAR ELECTRODES



RESPONSE FIELDS OF NEURONS EXCITED BY INTRACOCCHLEAR ELECTRODES

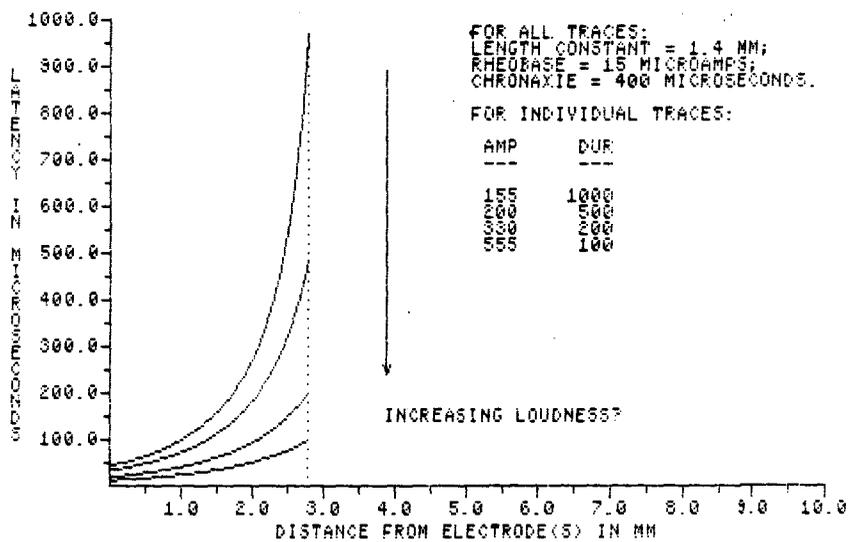


Fig. III.A.5. Top panel: Illustration of response fields produced by constant-charge pulses. Bottom panel: Illustration of response fields produced by stimuli that maintain a constant extent of the excitation field. Note the distinct changes in the synchronicity of modeled discharge for the response fields in the bottom panel.

"constrained" positions, the spatial resolution (and interactions between adjacent channels) of the presentation could be improved. Also, if both the extent of excitation and the synchronicity of input have effects on loudness, then both can be used to increase the number of discriminable steps of loudness perception. Specifically, if we can code steps of loudness with manipulations in synchronicity for every useful step of loudness coded by extent of excitation, then we can increase the number of steps of loudness coding beyond the number possible for stimulus schemes that do not exert independent control over these parameters of excitation.

The idea of coding loudness by manipulation of latency fields is further illustrated in Fig. III.A.6. Here, the waveforms of the stimuli delivered to intracochlear electrodes are shaped to produce "flat" latency profiles across the excited patch of neurons. If our hypothesis of loudness coding is correct, that both extent of the excitation field and synchronicity of input contribute to the percept of loudness, then percepts produced by such flat latency fields will be louder than the percepts produced by the curvilinear latency fields of Fig. III.A.5.

To test these hypotheses of loudness coding, we plan to measure the strength-duration characteristics of the neural population in the vicinity of each electrode pair in implanted patients using intracochlear evoked potentials (see section II.A)*. Then, with our knowledge of the field patterns produced by the UCSF electrode array (see section II.B) we can approximate the various conditions illustrated in Figs. III.A.5 and 6. Loudness matching and "same-different" paradigms will be used to evaluate the effects of substantially independent manipulations in the extent of the excitation field and in the synchronicity of discharge over this field. The data from the psychophysical tests should be adequate to assess at least crudely the relative contributions to loudness made by these manipulations. If synchronicity of excitation has an effect, then we will have an opportunity to increase the number of discriminable steps along the scale of

*Such direct measurements are required to so characterize these neurons because psychophysical threshold measurements reflect the central integration of peripheral inputs. In general, data from implant patients suggest that apparent "chronaxies" derived from psychophysical threshold curves may be substantially greater than the actual chronaxies of peripheral neurons.

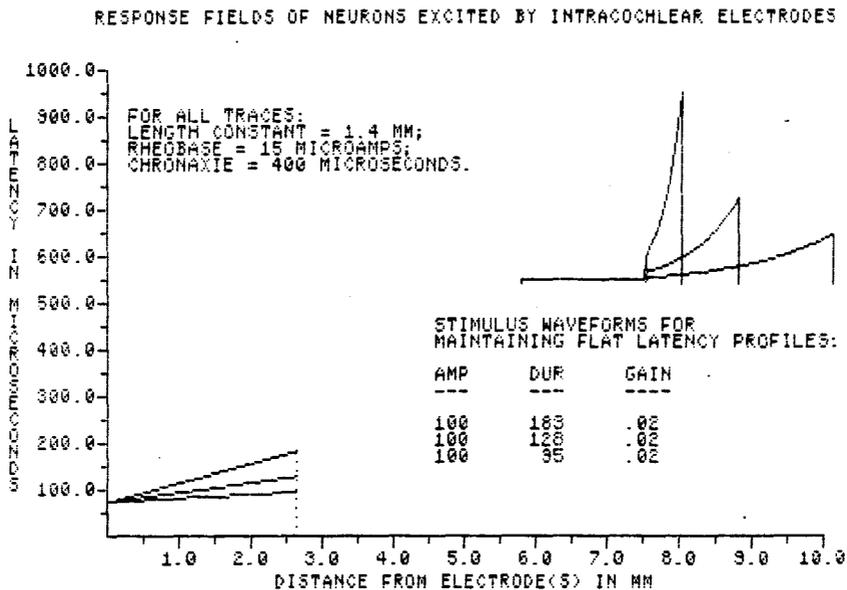
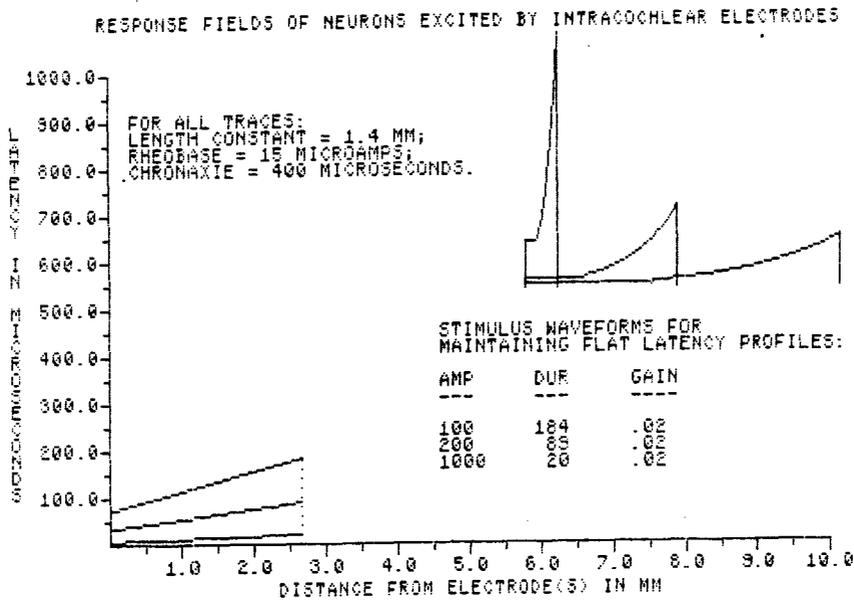


Fig. III.A.6. Illustration of stimulus waveforms required to produce flat latency fields within a restricted region of excitation. Top panel: Waveforms with various initial amplitudes (upper right; see table for waveform parameters) required to produce flat latency profiles with various absolute latencies at the position directly over the electrode(s). Bottom panel: Waveforms with the same initial amplitude required to produce flat latency profiles with the same absolute latency at the position directly over the electrode(s).

loudness and to improve the spatial resolution of the presentation, as previously described.

A final note in this subsection on loudness coding relates to the stimulus waveforms used to produce approximately-flat latency profiles across the field of excitation. Obviously, one would not want to have to generate such complicated waveforms at the outputs of speech processors for auditory prostheses. However, as Fig. III.A.7 shows, close approximations to the desired result of producing flat latency profiles can be achieved with rising-ramp stimuli. Falling-ramp stimuli can also be used to produce a different latency profile with essentially the same extent of excitation. As long as the limits for safe levels of maximum current are not exceeded, such stimuli might have useful applications in auditory prostheses. Also, these stimuli can be used in psychophysical tests to confirm the findings of our more-complicated studies described above (which involve direct measurements of strength-duration curves and some assumptions about the electric field patterns produced by the UCSF electrode array). That is, if the loudness of suprathreshold rising ramps is the same as the loudness of falling ramps of the same charge, then the hypothesis of synchronicity of input, as stated above, will not be supported.

3. Manipulation of latency profiles to code pitch

Much recent evidence suggests the possibilities that the frequencies of sounds may be encoded at the normal auditory periphery by (1) large discontinuities in the latency fields of neurons that correspond in best (or "characteristic") frequency to the frequencies of components in the stimulating sound, produced by the rapid accumulation of phase lags ("phase shear") in the terminal regions of basilar-membrane displacements (see Allen, 1983; Loeb et al., 1983; Shammé, 1985) and (2) coincidence of temporal inputs from widely-separated segments of the cochlear partition, that could code frequency by the distance along the basilar membrane between the coincident inputs (Loeb et al., 1983). Both of these mechanisms of pitch perception have been proposed to address the problem of how a constant percept of pitch can be maintained over the very wide dynamic range of intensities for normal hearing when even moderately-intense sounds excite a broad extent of neurons along the cochlear partition. Synchronicity of "phase-locked" inputs, sharpened by the mechanism of "synchrony suppression"

NEURAL RESPONSE (MAX LATENCY = 1000.0)
RESPONSE FIELDS TO RISING (LONGER LATENCY) AND FALLING RAMPS
(RAMP DURATION = 500 MICROSECONDS; MAXIMUM INTENSITY = 500 MICROAMPS)

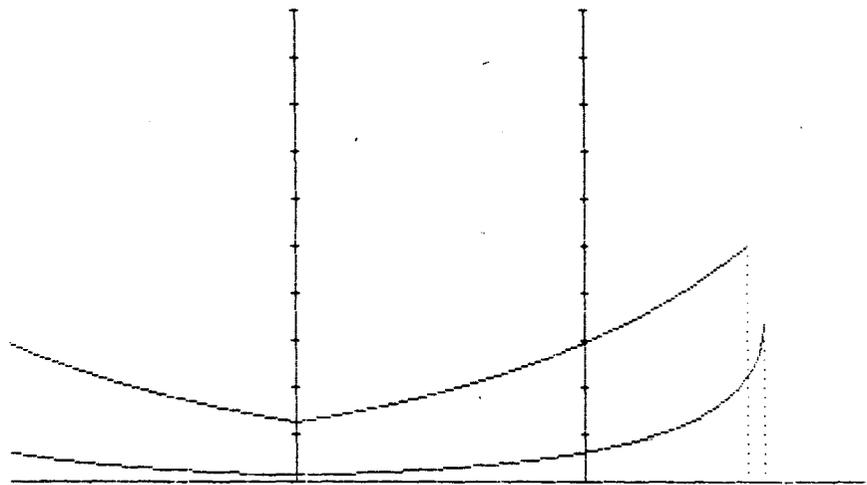


Fig. III.A.7. Neural response fields produced by rising and falling ramps. The vertical lines mark the positions of two electrode pairs in the UCSF array, which are spaced at 2 mm intervals. The graduations along the vertical lines mark 100 usec steps in "latency" after the onsets of the stimuli. The responses evoked by rising ramps have greater absolute latencies than the responses evoked by falling ramps, and the extents of the excitation fields produced by both are approximately equal.

(Sachs, 1984), phase shear and cross-correlation of widely-spaced inputs are all preserved over a much broader dynamic range than is the average rate profile of the responding ~~central~~^{neural} population. ~~Therefore~~^{Therefore} synchronicity, phase shear and "spatial cross-correlation" are potentially robust representations of frequency that could be used by the central auditory system to infer the psychological attribute of pitch. Neural mechanisms for central "decoding" of these representations of frequency have been proposed (Loeb et al., 1983; Sachs, 1984; Shamma, 1985). If one or more of these representations is in fact utilized by the central auditory system to infer pitch, then the deficits of pitch perception found for implant patients are not at all surprising. That is, the fine temporal details of these three representations are not reproduced in the patterns of neural discharge evoked by the electrical stimuli used in present auditory prostheses. In this subsection we describe ways in which latency profiles can be manipulated to produce approximations to the representations of phase shear and of coincidence of inputs from widely-spaced segments of the cochlear partition. In the subsection on "noise biasing" we describe a way in which synchronicity of phase-locked inputs might be approximated.

Coding of phase shear requires the production of sharp discontinuities in the latency fields of the responding population of neurons. Such discontinuities can be produced by a combination of the following: (1) manipulation of stimulus waveforms, much like that illustrated in Fig. III.A.6; (2) appropriate multichannel stimulation; and (3) appropriate interchannel timing of the shaped stimuli delivered to adjacent electrodes. Multichannel stimulation is required because the latency fields produced by single-channel stimulation are approximately symmetric on either side of the electrode (see Fig. III.A.7). As might be appreciated from the above description, the procedures required to produce a good approximation of the representation of phase shear are complex. However, this representation can be mimicked at the cost of distorting to an even greater extent the representation of the rate profile found in normal hearing. Inasmuch as phase shear may be a much more salient representation of frequency than the rate profile, a series of experiments to evaluate pitch perception (including measures of frequency DLs, pitch, salience and pitch saturation) with an approximate representation of phase shear would appear to be worth the effort.

Coincidences of inputs from widely-separated segments of the basilar membrane can be modeled in an auditory prosthesis by delivering stimuli to different electrodes in the implanted array at the proper instants. The "proper instants," of course, are determined by the properties of basilar-membrane displacements in normal hearing. Therefore, we will need to implement at least a simple model of basilar-membrane displacements to determine the timing relationships of stimuli to be delivered to electrodes in the array for a given sound. The model of basilar-membrane displacements will not have to be very complicated because only a crude representation of inputs from different locations along the cochlear partition can be conveyed with the limited spatial resolution of intracochlear electrical stimulation.

4. Unidirectional coding of loudness

Another possible code for the frequencies of input sounds in normal hearing is the precipitous falloff in displacements of the basilar membrane beyond the point of maximum displacements (Evans, 1978). For sinusoids this "sharp edge" is maintained over a broad dynamic range of intensities, and thus could be "read" by the central auditory system to infer pitch. Also, as intensity is increased for such simple stimuli, more and more neurons are entrained at locations basal to the point of maximum displacements. This progressive "basal recruitment" of neurons could serve as a code for the intensity of the input sound, while the sharp edge of the falloff in basilar-membrane displacements could continue to code the frequency of the input sound. In this way a constant percept of pitch could be maintained across a broad range of loudnesses.

Approximate representations of unidirectional recruitment can be produced with auditory prostheses by making the electrode or electrode pair that corresponds to the position of the frequency to be coded a "reference" or "anchor" point. To code near-threshold sounds, only that electrode or electrode pair would receive stimuli; to code more-intense sounds progressively more electrodes in the basal direction would receive stimuli. In contrast to the more-or-less symmetrical growth of the neural excitation field produced by increasing the intensities of stimuli delivered to single electrodes (see top panel, Fig. III.A.4), this scheme would preserve the apical edge in the excitation field as required by this coding strategy.

Unfortunately, there are two lines of evidence to suggest that unidirectional coding of loudness may not be useful for auditory prostheses. First, as indirectly indicated above, sharp edges are not maintained in normal hearing for complex, multifrequency sounds of moderate and high intensities (except, perhaps, for a small population of units with low rates of spontaneous activity, see Sachs, 1984, for a full discussion), and therefore this representation in isolation of others is unlikely to be an adequate code for frequency and intensity in normal hearing. Second, and also mentioned above, is the fact that increases in the intensities of electrical stimuli delivered to intracochlear electrodes are often accompanied by increases in perceived pitch (Muller, 1983; Shannon, 1983a). If the central auditory system attends to the location of the apical edge in the response field to infer pitch, then one would expect decreases in perceived pitch for increases in stimulus intensities because the excitation field produced by electrical stimulation extends both apically and basally with such increases. Therefore, we think it is unlikely that unidirectional coding of loudness will be useful in isolation for auditory prostheses. However, its use could improve the representation of frequency and intensity when combined with other representations of these stimulus attributes. We will thus evaluate unidirectional coding of loudness in combination with some of the other coding strategies presented in this subsection on stimulus primitives.

5. Noise "biasing"

Another aspect of neural discharge in the normal auditory periphery that may have significance for cochlear implants is the stochastic independence of activity between adjacent auditory fibers (Johnson and Kiang, 1976). This independence may allow the transmission of volley information to the central auditory system for frequencies well above the maximum rate of single-fiber discharge, which is around 200 Hz for normal hearing. Indeed, spike-interval input from 10 stochastically-independent fibers innervating a small region of the cochlear partition is adequate to explain frequency resolution in normal hearing (Goldstein and Srulovicz, 1977; Srulovicz and Goldstein, 1983; Wakefield and Nelson, 1985).

Stochastic independence is manifested in the normal auditory system by the lack of correlation between spike trains of spontaneous activity in

fibers with the same best frequency. It is upon this base of spontaneous activity that stimulus-evoked activity is superposed. The spontaneous activity imposes different discharge histories of neurons innervating the same region of the cochlear partition; with these different histories some neurons will be "prepared to respond" to each succeeding phase of a sinusoidal stimulus while other neurons will not be so prepared. A trading of roles and "preparedness" between neurons allows a small subset of neurons out of larger population to fire on each phase of the stimulus. This trading of rôles is the basis for the "volley theory" of pitch perception, in which information on the frequency of an input sound is conveyed in volleys of discharge from separate subsets of the responding field on each stimulus phase.

A major difference in the discharge activity between deafened and healthy ears is that there is no spontaneous activity in deafened ears. As outlined above, it may be that such spontaneous activity, to produce different discharge histories among closely-spaced neurons, is necessary for conveying volley pitch information above the maximum frequency limit of single-unit discharge. Without stochastic independence between adjacent neurons, one might expect "rate-pitch" percepts to saturate at the maximum frequency of single-unit discharge, which, for electrically-evoked hearing, is around 400 Hz. This is very close to the limit of discriminable changes in rate-pitch percepts for implant patients. Therefore, a method to induce stochastic independence among neurons in the stimulated population might be useful for extending the saturation limit for rate-pitch percepts.

One such method would be to superpose "deterministic" stimuli on a continuous background of high-pass noise. The noise would act to mimic spontaneous activity in the normal auditory nerve, and its passband and level would be chosen to exploit to the maximum extent possible the differences among neurons in the population to produce at least some degree

of stochastic independence.* For example, the level of the noise might be set in the middle of the 4-to-1 range of thresholds for single units, as reported by van den Honert and Stypulkowski (1984). The objective would be to impose different discharge histories for closely-spaced units, and thus allow normal transmission of frequency information by the volley-pitch mechanism.

To evaluate the idea of noise "biasing" we will measure intensity and frequency DLs of pulses and other stimuli with and without noise. Some of the procedures to be used are indicated in section II.D, "Patient Tests."

6. Manipulation of stimulus waveforms to improve the spatial resolution of the presentation

Several manipulations of stimulus waveforms have potential for improving the spatial resolution of neural excitation with the UCSF array. An example is presented in section III.A.1, "manipulation of latency fields to code loudness." Other examples are presented in section II.B, and relate to specific properties of the intracochlear electric fields produced by various stimuli with the UCSF electrode array. One such property of particular interest is that, for ears in which the lateral-most nodes of peripheral dendrites are preserved, one phase of "monophasic" stimuli produces a more-selective field of neural excitation than the opposite phase. Thus, the spatial resolution of excitation can be improved in such a case by proper selection of stimulus waveforms. We will explore these manipulations and others to improve the already-good spatial resolution of the UCSF electrode array.

*Although single units in the AVCN exhibit phase locking to intracochlear electrical stimulation up to and beyond the normal limit of 3 kHz (White et al., submitted), no measures of stochastic independence between neurons in the excited population have been made. In general, moderately-intense electrical stimuli produce ~~a~~ deterministic and highly-synchronous patterns of responses in the neural field of excitation. It is highly likely, in our opinion, that the degree of stochastic independence between adjacent neurons in electrically-evoked hearing is not constant over the dynamic range of stimulation, and that the maximum independence attained in electrically-evoked hearing is much lower than the independence found in normal hearing.

B. Design of Single-Channel Coding Strategies

Although we hope (and expect) that we will be able to extend the boundaries of present psychophysical performance of implant patients with our studies on stimulus primitives, current hypotheses of potentially-useful strategies for coding speech with auditory prostheses must work within these boundaries if they are to have any realistic chance for success. Present knowledge on implant psychophysics has been summarized in this proposal in the introduction to section III.A and by Muller in recent publications (Muller, 1981 and 1983).

As stated before, the main objectives of our work on single-channel coding strategies are to (1) develop single-channel, extracochlear auditory prostheses that can be safely and efficaciously applied in infants and young children; (2) improve methods for coding speech signals that reflect the excitation of the vocal tract, to complement information available in lipreading; and (3) provide a "building block" for multichannel speech processors in which signals representing the excitation of the vocal tract are coded separately from signals representing the "short-time" configuration of the vocal tract. Our specific strategies for realizing these objectives are the following:

1. Coding of voice/unvoice boundaries by delivering randomly-spaced pulses during the input of unvoiced speech and periodic pulses in synchrony with glottal openings during the input of voiced speech;
2. Coding of speech sounds that have both voiced and unvoiced components by using a special technique in speech analysis to derive excitation waveforms, which involves manipulation of the "residual" error signal from linear-prediction analysis of speech;
3. Coding of the first formant of vowels by using more "analog-like" or "noise-biased" waveforms for electrical excitation;
4. Coding of both first and second formants by presenting a signal that corresponds to F_2/F_1 instead of just F_1 , the former of which

not only has the potential to transmit information about both formants but also has the potential to normalize these formant data across speakers.

In the following subsections we will briefly describe each of these strategies and our tentative plan for evaluating them.

1. Further development of strategies to code F_0 and voice/unvoice boundaries

As a starting ^{point}~~point~~ in our efforts to develop single-channel coding strategies, we will be building on the excellent work of the group at Guy's Hospital in London (see, e.g., Walliker et al., 1985; Moore et al., 1985). In contrast to the widely-applied House device, the active electrode for the prosthesis developed at Guy's Hospital is not implanted in the scala tympani but instead is attached either to the promontory or round window. Extracochlear placement, of course, minimizes the risk of surgery and leaves open the possibility of later intracochlear implants (e.g., with multichannel devices for children beyond age 4 to 6, or with improved electrode arrays for adults and older children). In addition, the scheme used for stimulus encoding for the Guy's Hospital device is probably more appropriate for conveying features of speech than is the House device. The House device presents a 16-kHz carrier, amplitude modulated by the raw speech wave, to the active electrode, while the Guy's Hospital device explicitly extracts voice pitch from the speech wave and presents electrical pulses in synchrony with glottal openings during voiced-speech sounds. (The Guy's Hospital device also has a "MAPITCH" facility that can be used for mapping the extracted F_0 down in frequency.) As demonstrated in the many papers from the Guy's Hospital group, comprehension of speech tokens jumps from 30% obtained with lipreading alone to about 65% when the prosthesis is activated and acts as an adjunct to lipreading. Performance of this simple prosthesis does not seem to depend on the extent and pattern of dendrite survival in the inner ear. Therefore, with certain improvements such a device could provide great benefit to (1) deaf adults whose pattern of dendrite survival is presumed to be poor and (2) deaf infants whose rapid growth of the bony cochlear capsule prohibits safe implantation of

intracochlear electrodes prior to 3 or 4 years of age.*

Our general plan for the development of single-channel auditory prostheses is to evaluate several strategies of speech coding that could substantially improve the performance of the Guy's Hospital device and then to implement the most promising of those strategies in portable, real-time hardware. Additionally, we plan to compare the performance of these new strategies with block-diagram-compiler simulations of the present processing strategy used in the Guy's Hospital device and the strategy used in the single-channel Hochmair device (Hochmair and Hochmair-Desoyer, 1985).

The first new strategy, to be simulated by us and evaluated by the collaborating psychophysical teams, involves extraction and representation of F_0 and voice/unvoice boundaries. As outlined in section II.E of this proposal, we have implemented one such processor using the "Average Magnitude Difference Function" (AMDF) to extract these parameters from speech in both quiet and noisy acoustic environments. Representation of voiced speech segments will include the delivery of periodic pulses whose pulse repetition frequency is directly related to F_0 , as in the Guy's Hospital device. Unlike the Guy's Hospital device, however, unvoiced speech sounds will also be represented. Here, we will either deliver randomly-spaced pulses (as suggested in the "future plans" sections of papers from the Guy's Hospital group) or low-level noise stimuli when unvoiced speech sounds are detected. Clear transmission and perception of voice/unvoice and

*We note that workers at UCSF have been evaluating the feasibility of safe implantation of intracochlear electrodes in children. Their preliminary findings indicate that the growth of the otic capsule is rapid during the first two years of life, and that this rapid growth presents several problems for safe and beneficial implantation of intracochlear electrodes. These problems include (1) relative movement between the electrodes and target neurons, which can affect the stimulus-response properties of the prosthesis; (2) relative movement between the antenna/receiver and electrode array, which can compromise the link connecting them; and (3) changes in the dimensions of the scala tympani and round window, which can provide a pathway for infection of the inner ear through the round window. Although the UCSF team is working to solve these problems (e.g. by developing methods to seal the round window), present results do not support the notion that intracochlear implants are safe for children below the ages of 3 to 4.

speech/silence boundaries would provide additional information that is generally not visible on the lips.

2. "Multipulse excitation" using the residual error signal from linear-prediction analysis of speech

One of the leading techniques of speech analysis is to model the transmission function of the vocal tract and then compute an error signal that corresponds to the difference between an input speech wave and the modeled speech wave. The error signal is used to adjust parameters in the model so that the modeled speech wave closely approximates the actual speech wave. When actual and modeled waveforms are closely matched, the parameters alone can provide a very good description of the actual speech wave. These parameters can be transmitted at much lower bandwidths than the actual speech wave, and this reduction in bandwidth forms the basis of linear-prediction vocoders.

In most implementations of vocoders, including linear-prediction vocoders, excitation of the vocal tract is modeled as either a train of periodic pulses for voiced-speech sounds or as wide-band noise for unvoiced-speech sounds. At the synthesis end of the vocoder channel a "voice-unvoice" switch selects the excitation signal according to properties of the modeled speech wave. The problem with this approach is that the resulting synthesized signal sounds unnatural and "buzzy." Also, consonants with mixed periodic and noise-like excitation are not well modeled.

A solution to this problem has recently been proposed and tested by B. S. Atal of Bell Laboratories (Atal, 1983). Instead of using an explicit voice/unvoice switch (and associated speech analysis to make the voice/unvoice decision), he suggests using a technique he calls "multipulse excitation" in which no a priori assumption is made about the nature of the excitation signal. In this "multipulse" model the excitation signal consists of a sequence of pulses for all classes of speech sounds, including voiced and unvoiced speech. The amplitudes and timing of the pulses are derived from a weighted error signal produced linear-prediction analysis of the input speech wave. In general, only a few pulses are required to generate the different types of speech sounds. Typically only 4 pulses are presented every 5 msec and these pulses have a relatively-limited dynamic range of intensities. The quality of speech synthesized with multipulse

excitation is much better than the quality of speech synthesized by the voice/unvoice switch strategy.

These attributes of the multipulse excitation signal--including pulsatile waveforms, limited dynamic range, modest temporal bandwidth, and faithful representation of the excitation of the vocal tract--make this signal particularly attractive for use in auditory prostheses. In addition, the analysis algorithm to derive this signal is generally simpler than algorithms designed to extract F_0 and voice/unvoice boundaries from the speech wave. For these reasons we plan to simulate this strategy of multipulse excitation for evaluation by the collaborating psychophysical teams. Signals delivered to the electrodes will be appropriately-processed multipulse signals, such as those shown in Fig. III.B.1.

3. Coding of the first formant of vowels by using more "analog-like" waveforms for electrical excitation

The demonstrations by Mark White that changes in F_1 are perceived in electrically-evoked hearing with a single channel of stimulation (White, 1983) and by the Hochmairs that speech intelligibility declines when energy between 900 and 4000 Hz is removed from the stimulus waveforms derived by their single-channel speech processor (Hochmair and Hochmair-Desoyer, 1985) both indicate the potential for conveying formant information with a single-channel auditory prosthesis. We will use results from our studies on stimulus primitives to increase the bandwidth of transmission, if these results suggest a way in which this can be done (one possibility is the use of "noise biasing", see section III.A.5). Additionally, we will suggest to the collaborating psychophysical teams that they conduct formal studies on formant detection with a single channel of electrical stimulation. That is, with the two "analog-like" processors just described, measure the DLs for formant frequency and formant bandwidth over the ranges for F_1 . Typical inputs to the speech processor for such experiments are shown in Figs. III.B.2 and III.B.3. In Fig. III.B.2 synthesized waveforms for relatively low formant frequencies are presented and in Fig. III.B.3 waveforms for relatively high formant frequencies ^{are presented.} The psychophysical task will be to detect differences in these waveforms for changes in formant frequency and bandwidth. The results of such experiments will provide quantitative guidance for the improved design of single-channel speech processors that

"shape" analog inputs for electrical stimulation.

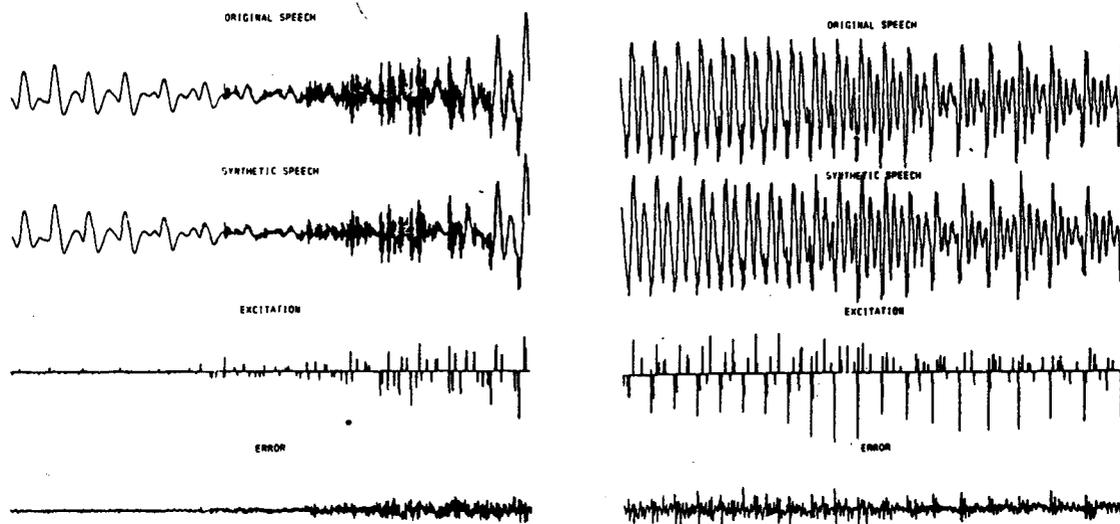


Fig. III.B.1. Two examples of original speech, synthetic speech, multipulse excitation, and error waveforms (from Atal, 1983).

STIMULI FOR F1-DETECTION EXPERIMENT, 50 HZ BANDWIDTH

F1 = 400 HZ



F1 = 500 HZ



F1 = 600 HZ



STIMULI FOR F1-DETECTION EXPERIMENT, 100 HZ BANDWIDTH

F1 = 400 HZ



F1 = 500 HZ



F1 = 600 HZ



Fig. III.B.2 Typical stimuli for an F1-detection experiment for the frequencies and bandwidths indicated. See text for details.

STIMULI FOR F1-DETECTION EXPERIMENT, 50 HZ BANDWIDTH

F1 = 900 HZ



F1 = 1000 HZ



F1 = 1100 HZ



STIMULI FOR F1-DETECTION EXPERIMENT, 100 HZ BANDWIDTH

F1 = 900 HZ



F1 = 1000 HZ



F1 = 1100 HZ

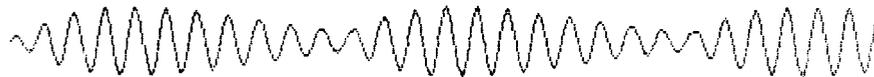


Fig. III.B.3 Typical stimuli for an F1-detection experiment for the frequencies and bandwidths indicated. See text for details.

4. Coding of both first and second formants by presenting a signal that corresponds to F2/F1 instead of F1 alone.

If we can demonstrate that F1 is well perceived over the range of F1 frequencies and bandwidths in the experiment just described, then we can realistically consider the possibility of coding both first and second formants by presenting a signal that corresponds to F2/F1 instead of just F1. This ratio would code formant transitions between consonants and vowels and has the potential to normalize these formant data across speakers (the latter of which may make it easier to "place" the F2/F1 signal within the perceptual space of the implant listener). The total information thus squeezed through a single channel of electrical stimulation, if decoded by the auditory system, would include that describing the excitation of the vocal tract as well as that describing the configuration of the vocal tract. Therefore, we may be able to convey nearly all parameters of speech that are essential for intelligibility through a single-channel auditory prosthesis. This, of course, would be a major advance.

C. Design of Multichannel Coding Strategies

Considerations that are relevant to the design of multichannel coding strategies have been outlined in previous sections of this proposal and are also presented in Appendix 4, "Two Hypotheses of Multichannel Speech Processing Schemes Based on Psychophysical Data from Implant Patients." In this section we will present a detailed outline of our present plan to develop and evaluate (with the assistance of the collaborating psychophysical teams) multichannel coding strategies. This plan includes efforts to

- (1) Characterize ^{the} "electrical-to-neural transformer" using measures of intracochlear evoked potentials.
- (2) Perform a benchmark comparison of results obtained with the present, analog UCSF processor and with the computer simulation of this processor.
- (3) Compare results obtained with variations in simulations of the basic UCSF processor; variations will include changes in compressor design, number of channels, configuration of the electrode array, and morphology of the stimulus waveforms.
- (4) Measure performance of the basic Australian strategy in the same patients.
- (5) Measure performance of variants of the basic Australian strategy, all designed to present information describing the excitation of the vocal tract "independently" from information describing the configuration of the vocal tract.
- (6) Measure performance of strategies that are intended to mimic patterns of "synchrony suppression" found along the array of VIIIth-nerve fibers for inputs of speech and other complex sounds in normal hearing.

- (7) Use results obtained in the characterization of the electrical-to-neural transformer (#1 above) to "optimize" the multichannel processors of points #3 - #6 above for the particular patient under test.

- (8) Measure degradations in performance of the best multichannel processors when interfering signals are added to the speech input; if the degradations are significant for one or more of the processors that work well without interference, evaluate techniques to improve their resistance to noise.

The block-diagram compiler will be used to simulate the processing strategies indicated above, and comparisons of different processing strategies will be made with single implant subjects, as previously described.

Tests to Evaluate Multichannel Coding Strategies

I. Characterize Electrical-to-Neural Transformer

- measure electrode impedance on all channels
- measure threshold for pulses and sinusoids on each channel (use CAP/EABR)
- measure channel interactions (use CAP and psychophysical techniques)
- use CAP techniques to determine
 - a) whether peripheral dendrites are present in the vicinity of the stimulating electrode(s) (by presence of low-threshold, .6 ms response, where the response is not masked by adjacent-channel stimulation);
 - b) the dynamic range of all responses;
 - c) the dynamic range of dendrite-only stimulation, if the dendrites are present as determined in test (a), with the aim of defining the range over which sector-by-sector control of the nerve can be effected;
 - d) strength-duration curves for the two populations (the separate curves will be obtained with subtraction measures).
- repeat these measures for monopolar configurations of the electrode array

(see next page for data on the waveform of compound action potentials (CAP's) obtained from intracochlear stimulation of the auditory nerve.)

- ### II. Perform a "benchmark" comparison of results obtained with the present, analog UCSF processor and with the computer simulation of this processor driving the electrodes via the DCU/Hardware Interface system. If the results are essentially identical, then we will have confidence in the results obtained from other tests in which the power of the computer-based system is exploited.

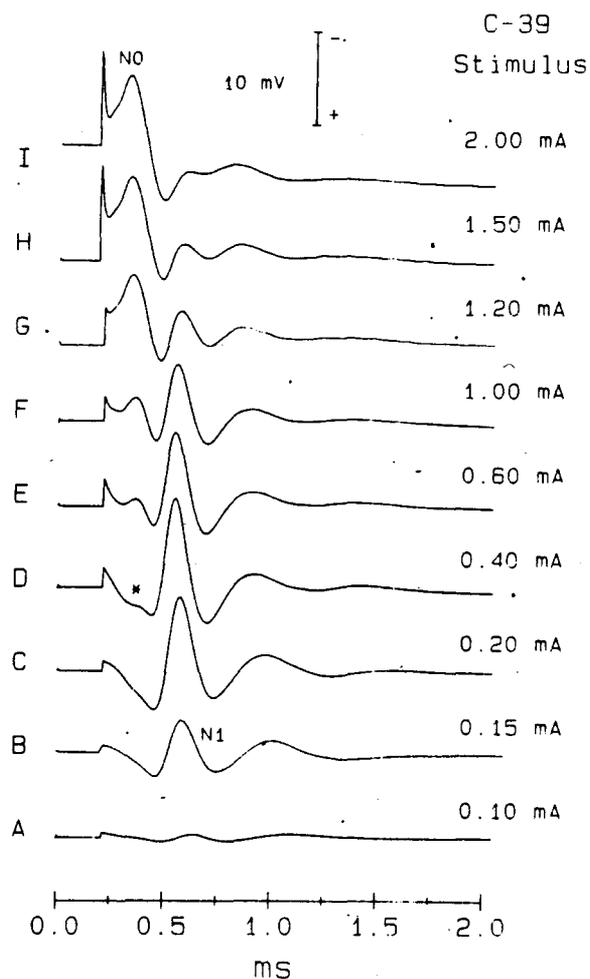


Fig. 1. Compound action potential recruitment. CAPs recorded from monopolar intracranial electrode. Stimuli were 100 μ s monophasic pulses applied to monopolar, intracochlear electrode. Stimulus intensity is given to the right of each response. At low stimulus strengths the N_1 component is most apparent (A-D). At moderate stimulus amplitudes the N_0 component (*) appears. At the highest stimulus amplitude the N_0 component dominates and N_1 disappears (I).

Figure 1. Waveforms of compound action potentials evoked by intracochlear electrical stimulation. The data are from P. H. Stypulkowski and C. H. van den Honert, "Physiological properties of the electrically stimulated auditory nerve. I. Compound action potential recordings," *Hearing Research*, 14 (1984) 205-223. Evidence presented in this paper strongly suggests that the N_1 component arises from stimulation of peripheral dendrites and that the N_0 component arises from stimulation of central axons.

III. Compare results obtained with variations in the design of the basic UCSF processor. Use the computer simulator to implement the different designs. Variations should include at least the following:

- changes in compressor design (e.g., multichannel vs. "front end"; linear vs. nonlinear; instantaneous vs. noninstantaneous; etc.)
- changes in the number of channels (suggest 1, 4, 8)
- changes in electrode configuration (e.g., monopolar vs. bipolar; changes in electrode channel assignments for bipolar arrays of less than 8 channels)
- changes in stimulus waveforms and timing to exploit knowledge obtained in the studies of "stimulus primitives" (the objectives of waveform manipulations will be to (a) improve the temporal and spatial resolution of the presentation; (b) expand the dynamic ranges of intensity and frequency coding; (c) preserve a constant percept of pitch in the face of coded changes in loudness; and (d) increase the salience of pitch percepts.)

IV. Measure performance of the basic Australian strategy in the same patients. Use the computer simulator to implement this strategy for eight (rather than 15 or 22) channels of stimulation. Include all coding limitations of the Australian strategy, some of which are imprecise (and sometimes grossly inaccurate) extraction of speech parameters, an inadequate number of such parameters to convey fully-intelligible speech, restricted bandwidth of transmission from the processor to the electrodes, and lack of stimulus waveform control.

- V. Measure performance of improved strategies for presenting information describing the excitation of the vocal tract "independently" from information describing the configuration of the vocal tract. Use the computer simulator to implement the tests. The tests should include the following:
- measure the performance of the basic Australian strategy, but use a processor in which voice pitch, V/UV boundaries, and F2 are accurately extracted and presented. Candidates for extraction algorithms include LPC analysis to extract F2; the Gold-Rabiner, SIFT or Cepstrum algorithms to extract voice pitch; and the Knorr V/UV detector to extract V/UV boundaries.
 - Use the algorithms listed above to extract and present F1 in addition to F2, voice pitch, and V/UV boundaries;
 - Use the LPC residual signal for excitation of individual channels, rather than trains of pulses corresponding to the voice-pitch signal (the idea here is to code more elegantly V/UV and silence boundaries, improve the coding of phonemes with mixed voice and unvoice components, and to simplify the processor by eliminating hardware for extracting voice pitch and V/UV boundaries)
 - exploit knowledge obtained in tests of stimulus primitives, again with the aims of improving the temporal and spatial resolution of the presentation; expanding the dynamic ranges for coding intensity and frequency; preserving a constant percept of pitch in the face of coded changes in loudness; and increasing the salience of pitch percepts.
- VI. Measure performance of strategies that are intended to mimic patterns of "synchrony suppression" found along the array of VIIIth-nerve fibers for inputs of speech and other complex sounds in normal hearing.
- (See next page for an illustration of synchrony suppression in the responses of single fibers in the auditory nerve to speech stimuli and for an example of a relatively-simple strategy to mimic the patterns of these responses in a multichannel auditory prosthesis.)
- VII. Use results obtained in the characterization of the Electrical-to-Neural Transformer (sect. I) to "optimize" the multichannel processors of sects. III - VI for the particular patient under test. Compare results obtained with processors designed without knowledge of the electrical-to-neural transformer to the results obtained with this knowledge. If these two sets of results are substantially different for individual patients, then devise and evaluate procedures for "prescribing" a particular multichannel processor (and parameters of that processor) for each patient.

VIII. Use the computer simulator of speech processors and the information obtained in sect. I to model the patterns of neural discharges elicited across the array for various speech and other inputs. The "neural transformer" part of the model will be similar to a model we already have, "MELECS" (for "multichannel electrical stimulation"). The integrated model will provide a tool for understanding psychophysical results in terms of discharge patterns (and their attributes of temporal bandwidth, spatial resolution, dynamic range, etc.). The model will have precisely the same inputs that are delivered to the electrode array in implanted patients. Finally, the model will be refined (a) by correcting deficiencies in its ability to predict psychophysical results and (b) by including ever-more sophisticated simulation algorithms from our field-mapping model.

IX. Measure degradations in performance of the best multichannel processors when interfering signals are added to the speech input. The interfering signals will include (a) broadband noise, (b) speech-spectrum noise, (c) speech babble, and (d) reverberant speech babble. If the degradations are significant for one or more of the processors that work well without interference, evaluate techniques to improve their resistance to noise. Candidate techniques include spectral subtraction, comb filtering, use of phased or directional microphones, and dereverberation.

D. Experimental Plan

Comparative evaluation of speech processors ^{for}~~or~~ auditory prostheses is a complex undertaking. The terrain is loaded with pits into which the unwary can fall. Therefore we will offer a few suggestions for the collaborating psychophysical teams while realizing that they will have the primary responsibility for conducting the evaluation studies.

First, we would like to suggest that there is a logical order in which the experiments should be conducted; that is, tests to characterize the "electrical-to-neural transformer" (see the first heading in the outline of tests presented in section III.C) and to evaluate stimulus primitives should be conducted before the tests to evaluate the performance of speech processing strategies. The rationale of this order of tests is simple: knowledge of the quality of the electrode/nerve interface and of perception of stimulus primitives is likely to guide the evaluation of speech processing strategies down the most productive paths. This is important because there are so many reasonable possibilities to consider in the evaluation of speech processing strategies. If some of these possibilities can be eliminated in the initial tests that precede those using speech materials, then time can be saved for evaluating the remaining possibilities in greater detail.

Next, when we get to the evaluation of speech processing strategies, it is obvious that the number of variable parameters will be very large (even with a reduction of possibilities in the preliminary tests) and therefore that only a limited number of variations can be entertained for each patient. The approach we suggest is to evaluate in single subjects the relative performance of the distinctly different classes of processors outlined in section III.C. The parameters of each of these processors should be set according to our "best estimates" from the results of the studies with stimulus primitives and intracochlear evoked potentials. Although a risk is taken with this approach of eliminating a good processor whose parameters are not adequately "optimized", a larger risk of not evaluating distinctly different strategies is avoided.

Tools for the evaluation of speech processing strategies should include those already used by the UCSF team. These include the "miniMAC" test, the "MAC" test, the Klatt synthesizer, and confusion matrices. Application of these tools has recently been reviewed by Ochs et al. (in preparation).

Finally, we would like to mention that the block-diagram compiler provides a powerful facility for not only randomizing the presentation of processed speech tokens, but also randomizing processing strategies within a single testing session. This feature could greatly improve the reliability of results obtained from studies to compare the performance of different processing strategies with single implant subjects.

E. Further Development of Portable, Real-Time Processors

As mentioned in the introduction to section III, our plan for further development of portable, real-time speech processors for auditory prostheses emphasizes the initial use and enhancement of components of hardware systems we have already designed and debugged at RTI (see section III.E.2 below). In this way we can bring to this project the benefits of considerable resources that went into the design and construction of these existing systems (see section III.E.1 below). For this "expanded scope" project we ^{also} propose to evaluate additional alternatives for processor designs and, on the basis of an initial evaluation, construct a device using the best suited state-of-the-art microprocessor available (see section III.E.3 below).

1. Instruments Based on the 80C31 and 80C88 Microprocessors

We have two basic designs of microprocessor-based, battery-operated instruments that are highly (and perhaps most) appropriate for the present applications. The first design is based on the CMOS version of the INTEL 80C31 microcontroller and is described in some detail in section III.E of this proposal. Briefly, this system has enough "computational horsepower" to implement most of the single-channel coding strategies outlined in section III.B and to act as a "building block" for multichannel processors that code information related to the excitation of the vocal tract separately from information related to the configuration of the vocal tract (see section III.C). The power consumption of our present hardware prototype using this design is very low, around 70 mW. The size and power consumption of the 80C31-based processor easily meet the demands of portability.

For more complex processors, of the type that will generally be required to implement the multichannel coding strategies outlined in section III.C, we have a hardware design for instruments based on the 80C88 microprocessor. This design has been developed and tested for use in our Autocuer (a portable, real-time, speech-analyzing aid for the deaf to disambiguate the ambiguities of lipreading) and in our personal monitor of cardiac function. Both instruments require small size, high digital throughput, significant memory, low consumption of power, and a well-behaved interrupt structure for sampling analog inputs. Extensive use of CMOS components is made in these instruments to reduce power consumption and

improve immunity to noise. The total power consumption of the Autocuer is around 700 mW and the total power consumption of the personal monitor is around 500 mW; these figures include the power consumptions of the front-end and display components to be described below. A 5.5-ounce rechargeable battery pack is used for both instruments and will power each for 10 continuous hours of use between recharges. Finally, both instruments fit into a 15.5 x 9 x 4.5 cm package when Multiwire (TM) circuit boards and "stacked memory" are used.

The front end of the Autocuer is an Interstate Electronics switched-capacitor filter bank (ASA-16), which has 16 2-pole (per skirt) bandpass filters, roughly spaced on a logarithmic scale of center frequencies. Each of the filter outputs is fed to a precision rectifier and the output of the rectifier is low-pass filtered by a 2-pole Butterworth filter whose break frequency is 30 Hz. This device is essentially a channel vocoder on a chip, and is used in conjunction with software in the Autocuer to extract accurately the frequencies of the first two formants (F1 and F2) of running speech.

The 80C88 is programmed to make decisions about the classes of speech sounds using the filter-bank outputs and digitized segments of the raw speech input. Once these decisions have been made, the processor relays this information to the user via LED displays mounted in a pair of eyeglasses. Tests with deaf subjects using simulated Autocuer outputs suggest that the device, in conjunction with lipreading, will allow full decoding and understanding of conversational speech. Twenty portable units will be fabricated next year for field trials. A photograph of the prototype Autocuer with eyeglass display is presented in Fig. III.E.1 and a detailed report on the hardware and software design of this instrument is presented in Appendix 2.

As mentioned before, another use of our basic digital design is in the personal monitor of cardiac function. The front end of this device provides signals of transthoracic impedance to the digital processor. This front end duplicates in every detail the functions of the relatively huge Minnesota Impedance Cardiograph. The digital processor logs changes in cardiac function over time, and presents status information on a liquid-crystal display. Data read back from the memory by a large computer are then analyzed for significance of changes in cardiac function produced, for example, by controlled exposures to pollutants or by the stress of

controlled amounts of exercise.

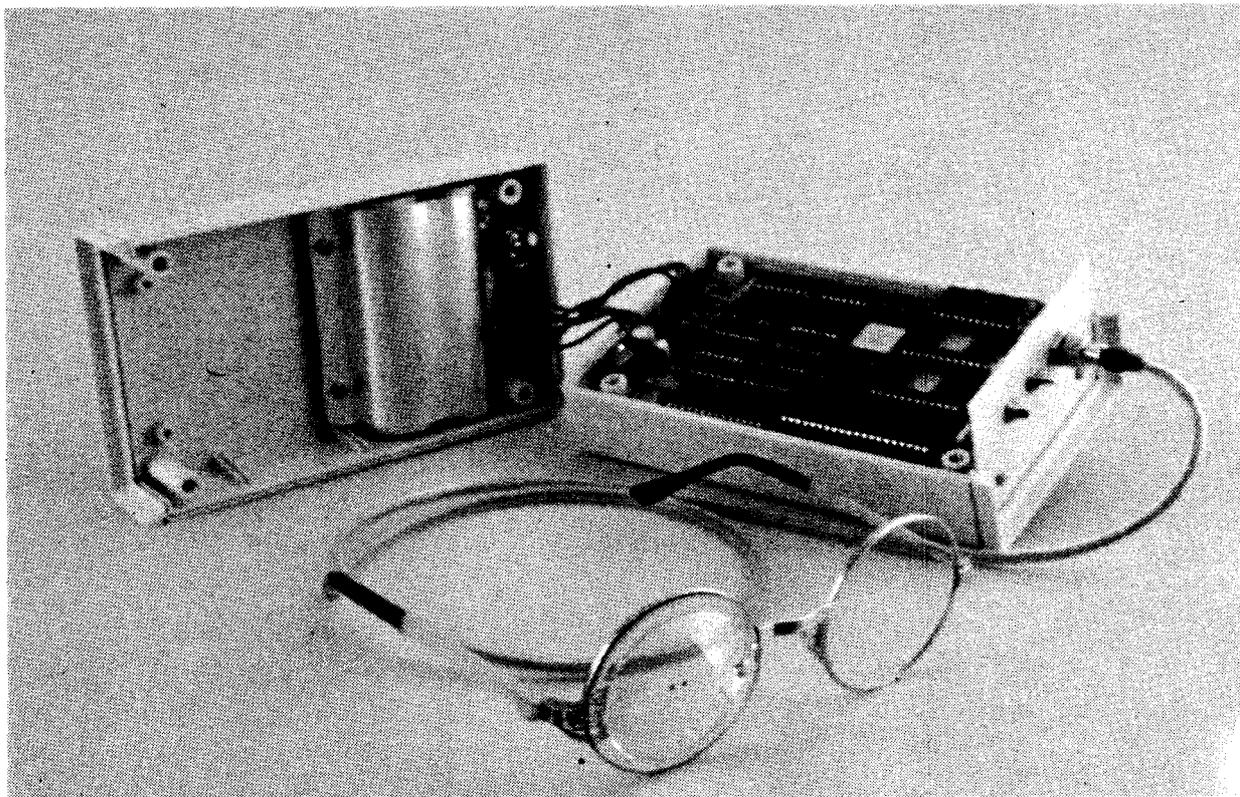


Fig. III.E.1. Hardware prototype of the Autocuer with eyeglass display.

Two views of the hardware prototype for the personal monitor are presented in Figs. III.E.2 and 3. This instrument is built on two Multiwire printed circuit boards, one containing the analog front end and the other

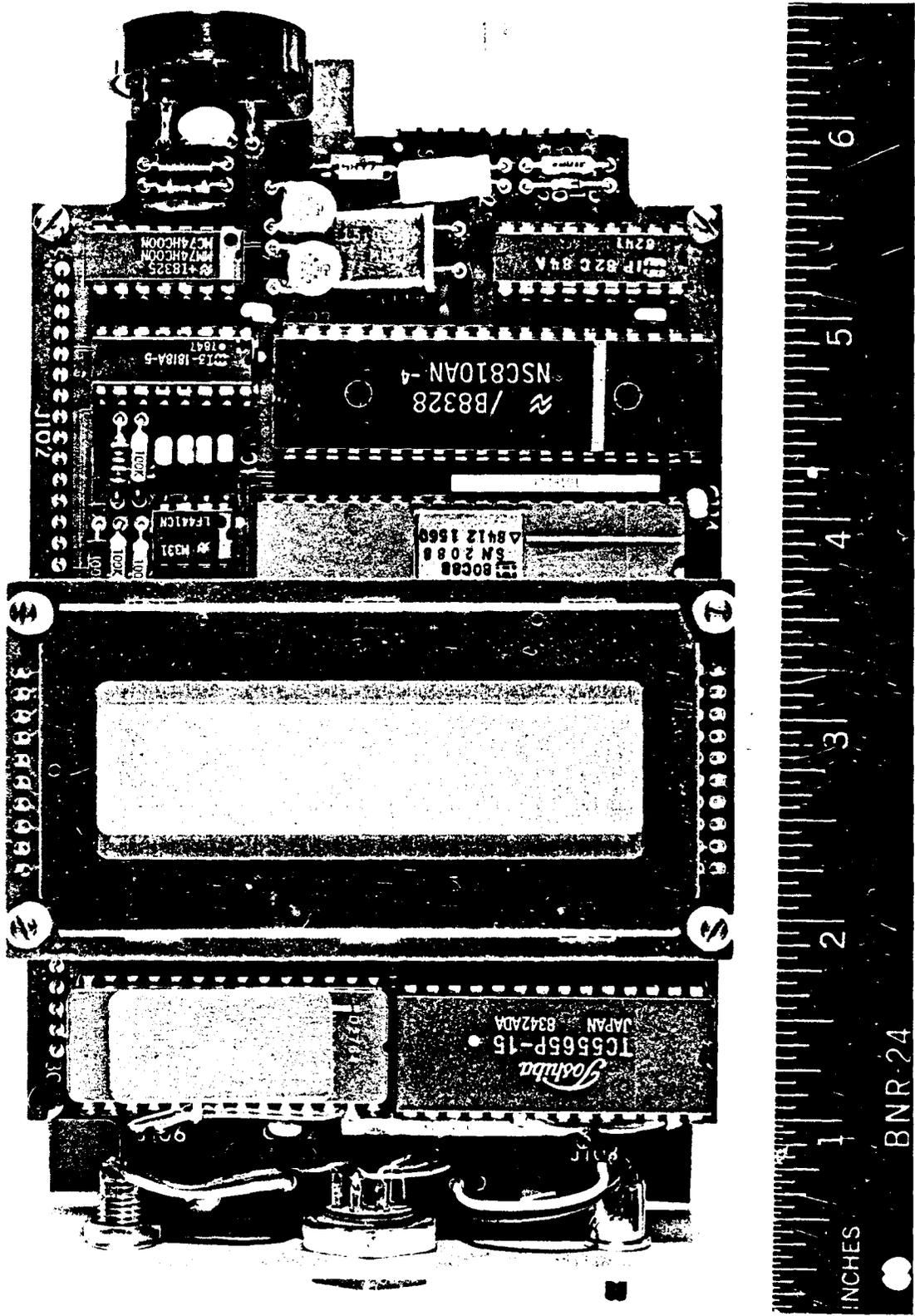


Fig. III.E.2. Top view of the hardware prototype of the personal monitor of cardiac function. The large component in the middle of the unit is a LED display.

the digital processor. The digital board is a complete CMOS microcomputer system capable of executing more than 500,000 instructions per second. The digital hardware complement includes 32 kbytes of CMOS memory (RAM plus EPROM), a 10-bit analog-to-digital converter with a maximum sampling rate of 10 kHz, two 16-bit timers, four parallel I/O ports, and an 80C88 microprocessor operating at 5 MHz. Including its rechargeable battery pack, the complete instrument weighs approximately 1.2 pounds.

An important advantage of the hardware approach for the digital processors just described is that the software for the system under development can be written and debugged using an IBM PC computer, which also uses an 8088-class microprocessor. This means that all the system and applications software for the PC can be used to produce error-free code for the portable processors. In addition, this means that we can use special-purpose software developed at RTI for portable devices that must in real time sample analog inputs, make complex decisions based on these inputs, and then convey the results to a display for output. Large "chunks" of code we have developed for other purposes could easily be adapted for use in a speech processor for a multichannel auditory prosthesis. For example, many hypotheses of speech-processor design suggest that accurate extraction of F1 and F2 will be required for subsequent representation of these essential speech parameters along the array of implanted electrodes. Such extraction could be performed using the Interstate Electronics filter bank and the software we have written for the 80C88 to interpolate its outputs. The accuracy of formant frequencies returned by the software routine is plus or minus 20 Hz, which is certainly adequate for coding of these frequencies in an auditory prosthesis.

2. Plan for Hardware Development

As mentioned before, we plan to use and build on components of hardware systems we have already designed and debugged at RTI. Further developments of the 80C31-based processor will include the following:

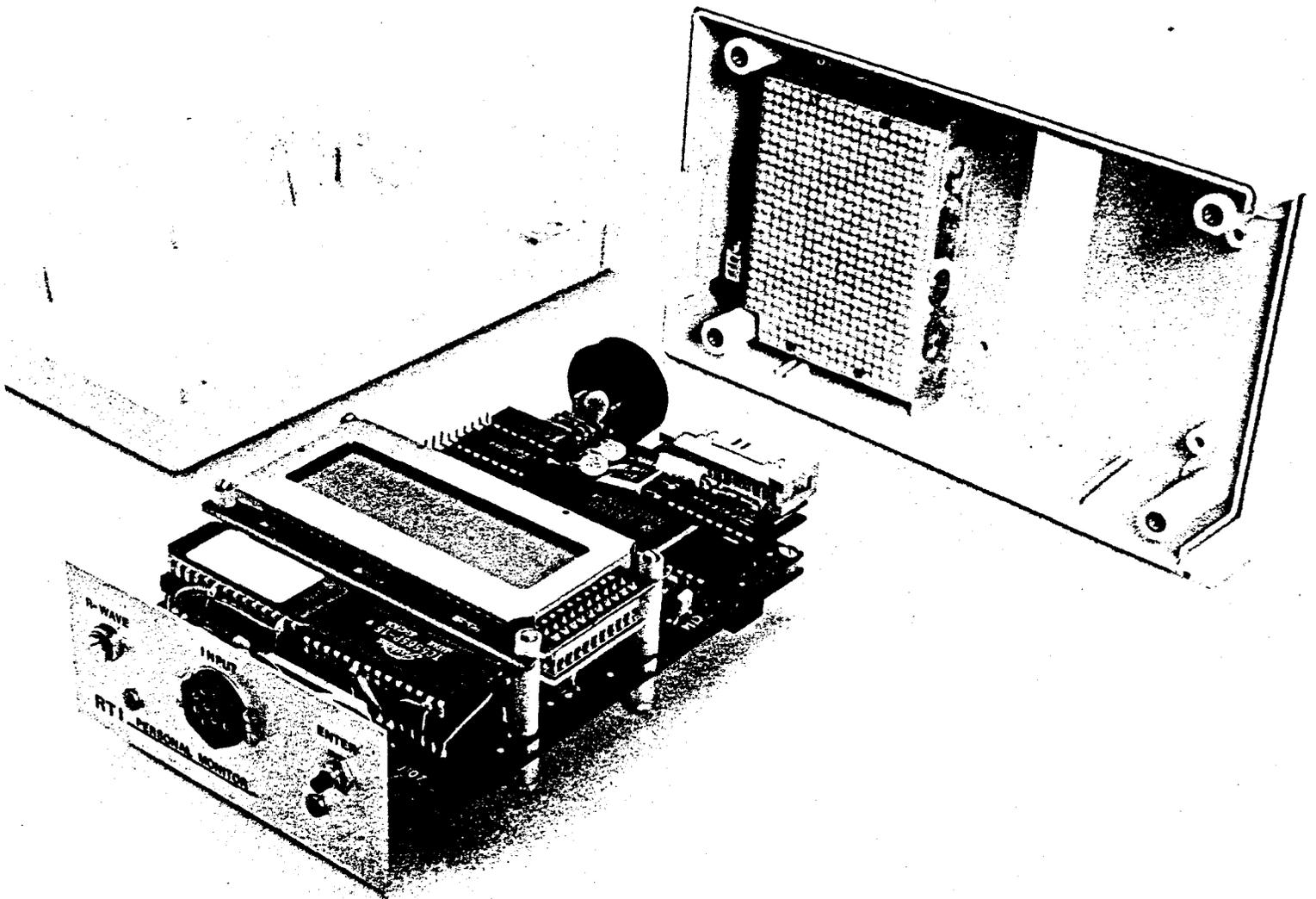


Fig. III.E.3. Angled view of the personal monitor with the top and bottom of the case shown in the background. The rechargeable battery pack fits within the top case, as shown on the right.

1. construction of prototype boards containing two 80C31s and one ADSP-1000 series multiplier, to increase greatly the computational capacity of this system without greatly increasing the consumption of power;
2. implementation of the "STREAK" algorithm with the present

processor with one 80C31, to derive and present the linear-prediction residual signal for single-channel auditory prostheses (see sections II.E and III.B);

3. implementation of a combined LPC residual signal and AMDF F_0 extractor with the processor with two 80C31s, to improve the accuracy of F_0 extraction (Un and Yang, 1977) and to provide alternate representations of signals describing the excitation of the vocal tract in one processor;
4. implementation of a computer-controlled gain stage in the analog section to keep signals within the dynamic range of the analog-to-digital converter;
5. implementation of more complex coding strategies, such as those that might be required for multichannel auditory prostheses, with the system containing two 80C31s and a multiplier.

These further developments can all be accomplished with a modest investment of engineering effort and will generally be directed toward improving portable processors for single-channel auditory prostheses.

Development of the 80C88-based instruments will be directed toward the objective of implementing in hardware the best strategies for coding speech with multichannel auditory prostheses, as determined from the tests described in sections III.C and D of this proposal. The present hardware system, with minor modifications, provides a substrate on which many of the coding strategies under consideration could be implemented. Because these strategies are so different in their approaches we can only give general indications (prior to our evaluation studies) of further developments we plan for the 80C88-based systems. These developments include:

1. Construct an optimized analog front end for an existing RTI wearable 80C88 processor to extract vocal tract information, utilizing the best available combination of commercial switched-capacitor filter banks and Microelectronics Center of North

Carolina custom integrated circuits.

2. Construct an analog-to-digital converter stage for the same existing processor design, with a sample rate appropriate to a speech processor for multichannel cochlear prostheses.
3. Design an output stage for wearable 80C88 processors to accommodate appropriate stimulus generation modules for intracochlear electrode arrays.
4. Implement promising multichannel processing strategies, as they are identified in the evaluation studies.

As with the 80C31 developments outlined above, these further developments of the 80C88-based processors can all be accomplished with a modest investment of engineering effort. The main tasks will be to redesign peripheral circuits and to adapt existing software, where possible, for the present applications.

3. Evaluation of Alternate Approaches to Hardware Design

Although we believe the approaches to hardware design just outlined are excellent ones for speech processors for auditory prostheses, we would be foolish to suggest that these are the only reasonable approaches for this application. Therefore, we are including in this "expanded-scope" proposal several promising alternate approaches to hardware design. In particular, we want to take a close look at the implications of using different microprocessors and of using integrated circuits specially designed for applications in digital signal processing (DSP).

Among the most attractive "DSP" chips are CMOS versions of the TMS320 and TMS32010 ICs made by General Instruments. These chips can implement in real time many of the signal processing functions that may be required in speech processors for auditory prostheses. These functions include ten-coefficient linear prediction analysis of speech (the results of which can be used to derive formant frequencies); determination of the fundamental frequencies of voiced-speech sounds; and more traditional DSP tasks such as

FFTs and autocorrelation of speech. These observations suggest that we may be able to simplify the design and reduce the power consumption of future speech processors for auditory prostheses by using the CMOS version of one of the TMS320 chips.

Another exciting possibility, only just recently announced, is the Motorola MC68HC11A8 advanced microprocessor, containing an impressive array of memory and peripheral features (including an 8-channel, 8-bit ADC) on a single chip with a maximum power dissipation of 110 mW. The cover sheet of the technical summary of this new device is reproduced as Figure III.E.4.

After an initial evaluation of these and other possibilities, we propose to build a processor using a development system based on the best currently available device for the purpose.

In exploring additional options for hardware designs, we shall be consulting closely with colleagues at the nearby Microelectronics Center of North Carolina who have expressed an interest in contributing effort toward production of custom CMOS ICs to support an effective ear-level processor.



MOTOROLA SEMICONDUCTORS

3501 ED BLUESTEIN BLVD., AUSTIN, TEXAS 78721

MC68HC11A8 Technical Summary

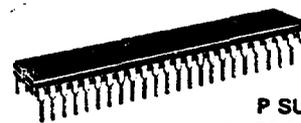
8-BIT HCMOS MICROCOMPUTER

The HCMOS MC68HC11A8 is an advanced microcomputer (MCU) containing highly sophisticated on-chip peripheral functions. An improved instruction set provides additional capability while maintaining compatibility with the other members of the M6801 Family. The fully static design allows operation at frequencies down to dc, further reducing its already low power consumption. Features include:

- 3 V to 5.5 V Operation
- Power Saving STOP and WAIT Modes
- 8K Bytes of ROM
- 512 Bytes of EEPROM
- 256 Bytes of Static RAM (All Saved During Standby)
- Enhanced 16-Bit Timer System
 - Four Stage Programmable Prescaler
 - Three Input Capture Functions
 - Five Output Compare Functions
- A Real Time Interrupt Circuit
- An 8-Bit Pulse Accumulator Circuit
- An Enhanced Non-Return-to-Zero Serial Communications Interface (SCI)
- A New Serial Peripheral Interface (SPI)
- Eight Channel 8-Bit A/D Converter
- A Computer Operating Properly (COP) Watchdog System
- Multilevel Interrupt Priorities (21)

HCMOS
(HIGH-DENSITY HIGH-PERFORMANCE
SILICON GATE)

MICROCOMPUTER

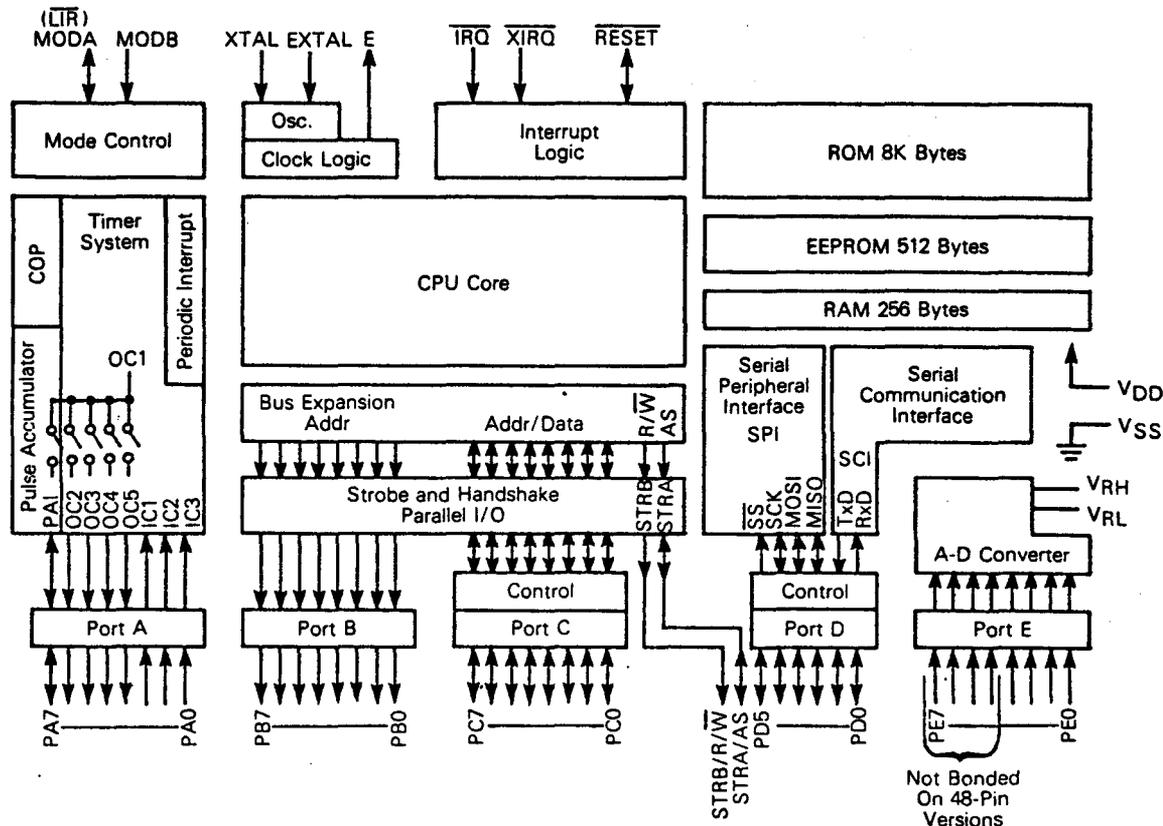


P SUFFIX
PLASTIC PACKAGE
CASE 767



FN SUFFIX
PLASTIC PACKAGE

FIGURE 1 — BLOCK DIAGRAM



F. Prospects

We expect to meet the major objectives outlined in this proposal. In particular, we think the prospects for conveying fully intelligible speech with multichannel auditory prostheses are good. We are sanguine about these prospects because results of recent tests in San Francisco (Ochs et al., in prep.) strongly suggest that (1) 100% "categorical" perception of changes in F2 is achieved when F2 crosses the boundary between two filters in the front end of the present UCSF speech processor; (2) with only four channels of stimulation significant recognition of speech tokens is achieved, including significant open-set recognition; and (3) when the assignments of channel outputs to electrode pairs are altered the pattern of errors in confusion matrices shifts in predictable and nearly "orthogonal" ways. These observations further suggest that, for patients with good nerve survival, use of all eight channels in the UCSF electrode array could greatly improve already-good levels of speech recognition. Possibly, if we have a patient with good nerve survival, then we may be able to convey fully-intelligible speech with a straightforward extension of the present UCSF processing strategy.

The possibility of meeting our primary goal, to define the classes and parameters of processor design that will allow full recognition of speech without lipreading for recipients of multichannel implants who have good survival of peripheral dendrites, is further enhanced by the fact that we will be evaluating (with the collaborating psychophysical teams) alternative strategies for coding speech with multichannel implants. As outlined in section III.C, several of these strategies are quite different from the present UCSF strategy. However, all the proposed strategies appear to have promise for conveying intelligible speech and ^{are} ~~is~~ firmly based on our present knowledge of implant psychophysics. Therefore, we believe one or more of these strategies is likely to emerge as highly useful for implant patients in whom ^{nerve} ~~nerve~~ survival is good.

We are also sanguine about the prospects for patients who have patchy or poor survival of dendrites in their inner ears. First, selected multichannel coding strategies are likely to work better for this class of implant patients than others and second, knowledge obtained in our studies of single-channel coding strategies is likely to be useful for the design of processors that convey at least some of the most-important elements of

speech to such patients. A particular prospect here is that we may be able to improve the representation of signals that reflect the excitation of the vocal tract, as described in section II.B. Inasmuch as this information is largely complementary to information available in lipreading, we may be able to restore to a useful degree speech recognition and perception in these patients.

Next, we hope that results of our work on single-channel coding strategies can be applied to the design of improved speech processors for extracochlear auditory prostheses. Our objective for these devices is to provide useful speech input to infants and young children at a time in life critical for the normal acquisition of language. As has been demonstrated by the UCSF team in recent studies, procedures for the safe implantation of intracochlear devices have not yet been developed for infants and young children.

Finally, we believe results of our studies on "stimulus primitives" will not only help to improve the performance of auditory prostheses, perhaps in very significant ways, but will also improve our understanding of the encoding of electrical and acoustic stimuli at the auditory nerve.

IV. Project Organization and Management

A. Personnel

A chart of the organization of the Research Triangle Institute (RTI) is presented in Fig. IV.A.1. The line of managers who would oversee the work proposed here is indicated in the figure by arrows. Responsibilities and time commitments of personnel involved in the technical aspects of the work are indicated in Table IV.A.1 for the first year of the proposed project and in Table IV.A.2 for the second and third years. The principal investigator will be Blake S. Wilson and working with him will be Charles C. Finley, Dewey T. Lawson, Warren J. Jochem, Robert L. Beadles, Phillip Rasberry and Kathrinn Fitzpatrick. Warren Jochem and Phil Rasberry are members of RTI's Center for Biomedical Engineering, under the direction of Robert L. Beadles. The only difference between Tables IV.A.1 and 2 is in the entries for Phillip Rasberry; his level of effort is greater in the first year so that he can build the hardware interface for use by the cochlear-implant team at Washington University and Central Institute for the Deaf (Wash. U./CID).

In addition to the above-named individuals, members of the cochlear-implant teams at the University of California at San Francisco (UCSF), Duke University Medical Center (DUMC) and Wash. U./CID will make substantial contributions to this project. The nature and extent of these contributions is fully outlined in section IV.C of this proposal, "Collaborative Arrangements."

RESEARCH TRIANGLE INSTITUTE
Officers and Research Programs

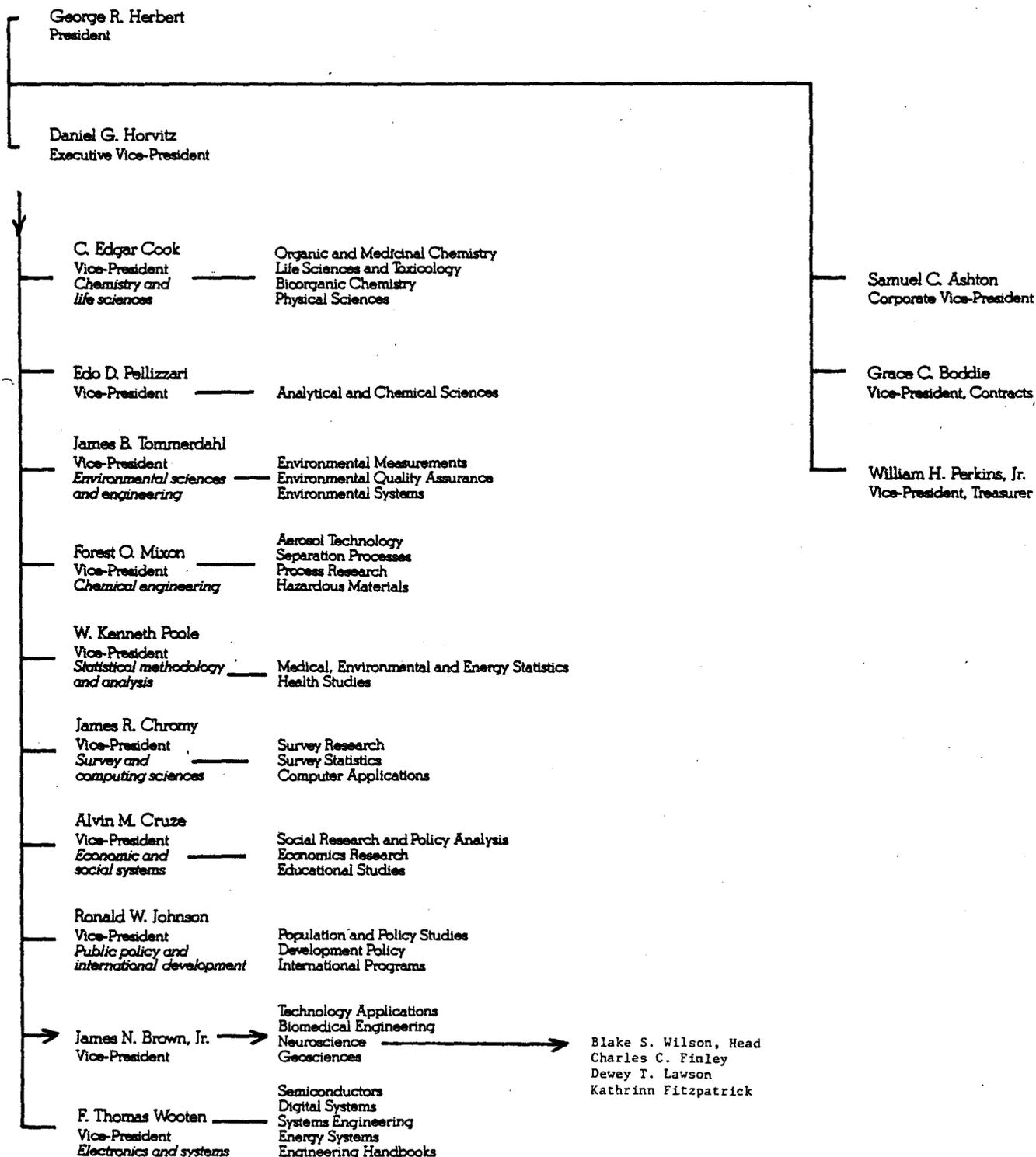


Fig. IV.A.1. Organization of the Research Triangle Institute.

Table IV.A.1. Responsibilities and time commitments of RTI personnel for the first year of the proposed project.

<u>name and title</u>	<u>responsibility</u>	<u>time</u>
B. Wilson, Head, Neuroscience Program Office	direct RTI effort as described in this proposal, the main elements of which are to design improved coding strategies for auditory prostheses; build tools for the evaluation of these strategies as applied in implant patients; and build portable, real-time processors to implement the best of these strategies in wearable devices	90%
C. Finley, Neuroscientist/ Research Engineer	participate in all aspects of the outlined project, with particular emphasis on work relating to refinements and engineering support of the hardware interface system; applications and limited further development of the integrated field-neuron model; design and preparation of software for implementing automated psychophysical tests; assist in the design of hardware for portable, real-time speech processors; support testing programs at the three collaborating institutions	90%

(see next page)

D. Lawson, Senior Physicist	support project activities in the areas of speech analysis; computer- based ^{based} management of speech data files; design and preparation of software for implementing procedures to obtain measures of speech understanding in cochlear-implant patients; design coding strategies for auditory protheses, especially those that use linear prediction analysis of speech	70%
W. Jochem, Senior Electrical Engineer	hardware design of speech processors for auditory protheses; evaluate alternatives to hardware design, including those suggested by use of the 80C31, 80C88, TMS32010 and MC68HC11A8 microprocessors	40%
R. Beadles, Manager, Center for Biomedical Engineering	hardware design, especially as it relates to the use of microprocessor technology in applications requiring real-time analysis of speech	5%
P. Rasberry, Electronics Technician	fabricate and test hardware prototypes; maintain computer equipment used in this project; build hardware interface for the St. Louis group	50%
K. Fitzpatrick, Secretary/ Administrative Assistant	reporting and correspondence; maintain records related to this project	20%

Table IV.A.2. Responsibilities and time commitments of RTI personnel for the second and third years of the proposed project.

<u>name and title</u>	<u>responsibility</u>	<u>time</u> →
B. Wilson, Head, Neuroscience Program Office	direct RTI effort as described in this proposal, the main elements of which are to design improved coding strategies for auditory prostheses; build tools for the evaluation of these strategies as applied in implant patients; and build portable, real-time processors to implement the best of these strategies in wearable devices	90%
C. Finley, Neuroscientist/ Research Engineer	participate in all aspects of the outlined project, with particular emphasis on work relating to refinements and engineering support of the hardware interface system; applications and limited further development of the integrated field-neuron model; design and preparation of software for implementing automated psychophysical tests; assist in the design of hardware for portable, real-time speech processors; support testing programs at the three collaborating institutions	90%
D. Lawson, Senior Physicist	support project activities in the areas of speech analysis; computer-based management of speech data files; design and preparation of software for implementing procedures to obtain measures of speech understanding in cochlear-implant patients; design coding strategies for auditory prostheses, especially those that use linear prediction analysis of speech	70%
W. Jochem, Senior Electrical Engineer	hardware design of speech processors for auditory prostheses; evaluate alternatives to hardware design, including those suggested by use of the 80C31, 80C88, TMS32010 and MC68HC11A8 microprocessors	40%
R. Beadles, Manager, Center for Biomedical Engineering	hardware design, especially as it relates to the use of microprocessor technology in applications requiring real-time analysis of speech	5%
P. Raspberry, Electronics Technician	fabricate and test hardware prototypes; maintain computer equipment used in this project	35%
K. Fitzpatrick, Secretary/ Administrative Assistant	reporting and correspondence; maintain records related to this project	20%

B. Facilities

Included in this section are brief descriptions of various resources available in support of the proposed work. The programs and capabilities of the Neuroscience Program Office are outlined on the following page, while a general description of Research Triangle Institute as a whole may be found in Appendix 5.

The facilities singled out for discussion in the subsections that follow are:

1. The standardized data processing architecture that allows RTI to develop software that is readily usable by all the collaborating medical centers.
2. Our unique hardware interface for delivery of stimuli to patients at each of the collaborating medical centers.
3. The RTI Center for Biomedical Engineering's facilities for development, construction, and testing of complex analog and microprocessor-based electronic instruments.
4. An existing, online database of digitized, segmented, and phonemically analyzed continuous speech, with a full array of supporting software utilities.
5. The National Biomedical Simulation Resource at nearby Duke University.
6. An extensive computer-based bibliographical and photocopy retrieval system for publications on cochlear implantation and related topics.
7. Additional computing resources being allocated to the Neuroscience Program Office by RTI.

**Programs and Capabilities
In Neurosciences**

Blake S. Wilson, Manager
Telephone (919 541-6974)

Has capabilities in neurobiology, neurophysiology, electrical engineering, and speech analysis. Collaborates with medical researchers, particularly at the Duke University Medical Center and the University of California at San Francisco.

Speech Processors:

Design processors to extract from speech parameters essential for intelligibility;
Encode these parameters for excitation of the auditory nerve on a sector-by-sector basis via implanted electrodes;
Develop a computer-based system for emulation of speech processors in software;
Develop speech processors to improve performance of hearing aids.

Neurophysiology:

Develop mathematical models of the vestibular system, and evaluate models in animal studies;
Investigate temporal processing mechanisms in the auditory system.

**Biological Effects of Microwave
Exposure:**

Identify effects of microwave exposure on brain activity in animals;
Record electrical activity in brain structures using single-unit techniques;
Record patterns of brain metabolic activity using radiolabeled deoxyglucose.

For clients in industry and government, Research Triangle Institute's staff of more than 900 conducts research in engineering, physical, life, chemical, social, and statistical sciences.

RTI is one of the nation's largest not-for-profit contract research organizations. It was established in 1958 as the cornerstone of North Carolina's Research Triangle Park, which is now the home of more than 45 industrial and governmental research centers. RTI often par-

ticipates in collaborative research with its founding universities—Duke University, North Carolina State University, and the University of North Carolina at Chapel Hill.

General information about RTI is contained in its annual report, which provides an overview of its organization and research programs. For copies, please contact the Marketing Support Office (919) 541-7044, or by mail at the RTI address.

1. Data Processing Installations

A key factor in our ability to conduct meaningful patient testing at multiple clinical sites is the availability, at RTI and at each collaborating medical center, of highly compatible computer hardware. To the extent such compatibility can be achieved, fewer program resources need be devoted to dealing with differences among the clinical sites.

Figure IV.B.1 represents the basic architecture of the computer systems used for patient stimulation at RTI, UCSF, and Duke. [It is not proposed that any patients be tested at RTI; the stimulator installation there is necessary for software development and evaluation.] The essential common elements are: (1) a central processor compatible with the Eclipse S/140 instruction set and equipped with either firmware or hardware floating point instructions and enough RAM and high speed disk memory to support Data General's Advanced Operating System (AOS); (2) a digital control unit (either a DCU/50 or a DCU/200) to support sustained high speed transfers of stimulus data; and (3) the parallel interface necessary to connect the DCU to the RTI Patient Stimulator. Other common features that, while not absolutely essential, are important in terms of efficient collaboration include: (4) industry standard magnetic tape drives to allow distribution of lengthy stimulus data files for patient testing; and (5) 1200 baud telephone line modems for distribution of new and revised source programs from RTI to the clinical sites and for software installation and testing by RTI via remote control.

In the figure, horizontal dashed lines indicate ways in which data can be exchanged between systems. In addition to the modes already discussed as common to all three installations, the two local systems at RTI and Duke also can exchange 5 Mbyte disk cartridges and diskettes.

Washington University in St. Louis recently has notified us of its intention to obtain and install, with assistance from the Central Institute for the Deaf, a fourth system meeting this standard, for use with an RTI Patient Stimulator in testing cochlear implant patients there.

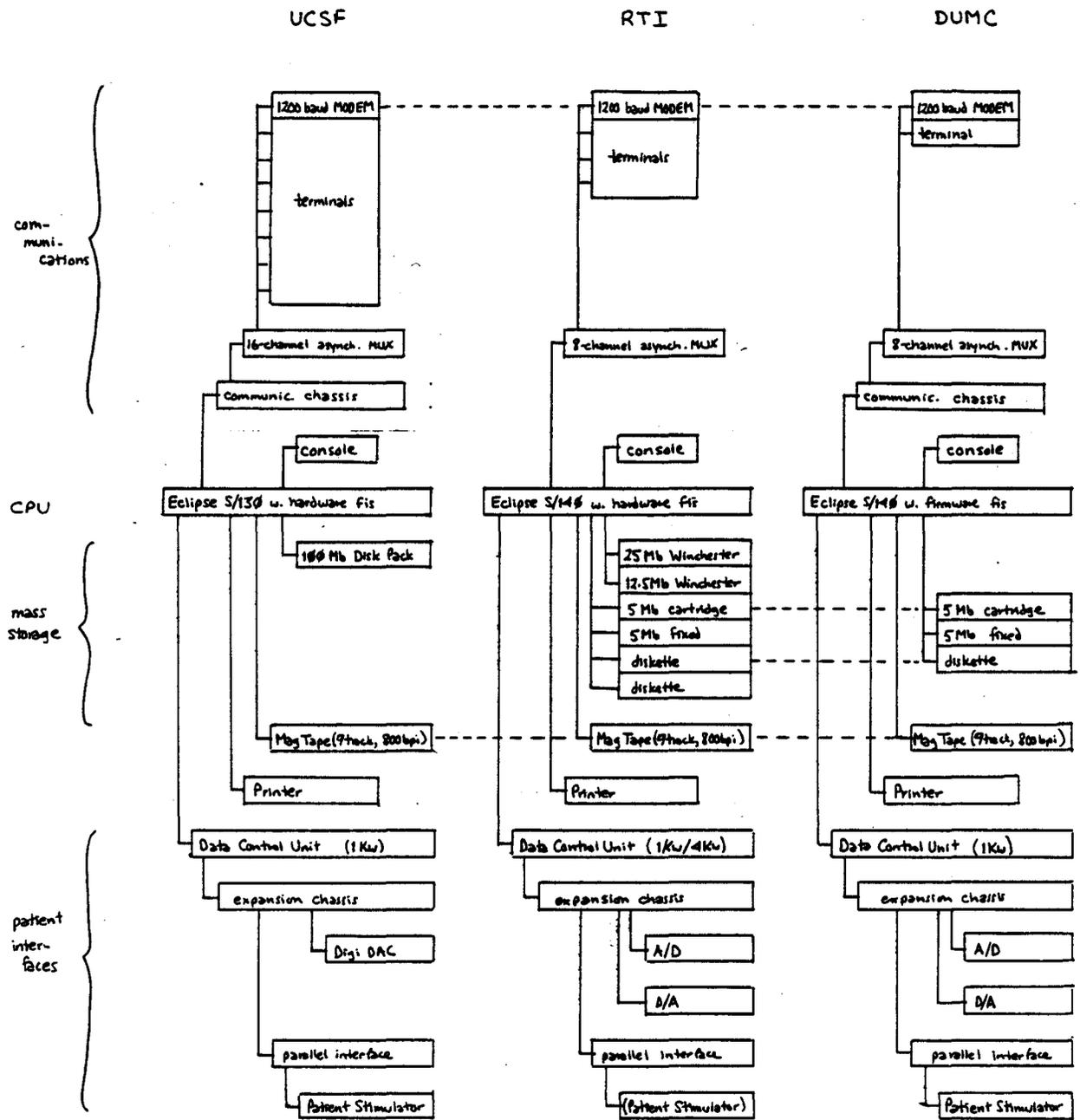


Fig. IV.B.1. Stimulus Processing at UCSF, RTI, and DUMC

2. The RTI Patient Stimulator

The unique combination of flexibility, bandwidth, and dynamic range provided by the RTI Patient Stimulator is crucial to much of the work we propose to do. This device will be available to each of the collaborating medical centers for patient testing. The large bandwidth and dynamic range allow the planning of a great variety of tests and the emulation of a broad assortment of processing strategies without encountering hardware constraints. The inclusion of response lines facilitates the automation of tests and the efficient use of limited time with each patient. The prospect of measuring intracochlear potentials in association with the stimulation of nearby electrodes holds enormous potential for improved patient assessment and processor design.

3. Facilities for Development of Biomedical Electronics

RTI's Center for Biomedical Engineering has substantial experience in developing advanced electronics systems for biomedical applications, particularly in the area of portable and wearable microcomputer-based systems. A wide range of laboratory electronic instrumentation, in-house parts inventories, and fabrication capabilities are available for rapid evaluation of candidate systems and for thorough shakedown of systems selected for full development.

The Center has extensive resources for research and development of microprocessor software and hardware. For physiological signal processing, an IBM Personal Computer (PC-XT) is available with 512K of read/write memory, 10 Mbyte fixed disk, 360K floppy disk, Data Translation DT2801 analog-to-digital subsystem (16 channel, 12-bit, 35 KHz analog-to-digital converter), TECMAR high-resolution graphic display (640 x 400 pixels, 16 colors), and Summagraphics MM1201 graphic digitizer.

For system development, three IBM-PCs and one PC/AT, all with hard disks and diskette drives, are configured as microprocessor development systems; one of these has RTI-developed real-time emulation capability for the Intel 8088 microprocessor. A Tektronix 7D02 logic analyzer is available for use alone or in combination with real-time emulation. MS-DOS, CP/M-86, and concurrent CP/M-86 operating systems are available on these computers, and PL/M-86, PASCAL, FORTRAN, BASIC and assembler languages are supported.

Each system has its own graphics printer and color monitor with graphics capability. Other language compilers, including ADA and C are available as needed.

The Center also has considerable expertise and experience in the area of electrical safety assurance in hospital and clinical research environments.

4. RTI Speech Database Facilities

An extensive speech database, available online through the Eclipse S/140 computer at RTI will help in several ways with the proposed work. The database is designed to incorporate speech data that differ in many respects (sample rate, amplitude precision, storage format, etc.) in a way that relieves the user of any need to deal with such complexities. Speech samples ranging from whole sentences through single words to small parts of individual phonemes can be obtained with great ease and flexibility. Much of the continuous speech in the database has been segmented and labeled using an RTI 60-phoneme analysis scheme. Multiple examples of a particular phoneme, for instance, can be located automatically and input to a particular processing algorithm under evaluation. The full range of software tools accompanying this database are available to aid in digitizing, editing, and organizing the speech samples needed in testing cochlear implant patients. Digitization of the MAC and MiniMAC tests is now in progress.

5. The National Biomedical Simulation Resource

The National Biomedical Simulation Resource at Duke University is available, without transfer of funds, for the execution of our integrated field-neuron model (see section II.B.1 above). This system consists of an AD-10 digital simulation computer (manufactured by Applied Dynamics International) connected to a VAX-11/750 host computer (Digital Equipment Co.). The AD-10 is a special-purpose, differential equation processor that can solve the Hodgkin-Huxley relations in real time. This speed of execution is about 300 times greater than that of our Eclipse computers. Thus, with a modest amount of programming effort to transform our present code for the system at Duke, we hope to enhance tremendously the utility of

our integrated field-neuron model.

6. Computer-based Bibliographical and Document Retrieval System

Bibliographic information and photocopy indexing for the Neuroscience Program Office's extensive library of publications on cochlear implants and a variety of related topics are made easily accessible to the professional staff through the use of SCIMATE database management software. This facility is used as well to maintain up-to-date documentation on the many computer programs constantly being generated and revised within NPO. Extensive use of key word labels on the entries to both these databases greatly enhances the accessibility and usefulness of the information they contain.

7. Additional Computing Resources Allocated by RTI

In addition to making an Eclipse S/140 computer system available to our collaborators at the Duke University Medical Center, RTI has agreed to augment the Neuroscience Program Office's facilities with the following capital equipment: (1) an IBM PC/AT computer with 1 Mbyte of RAM, 20 Mbyte disk, 1.2 Mbyte diskette drive, math coprocessor, enhanced color graphics, and local network hardware and software; (2) a Data General One "lap top" computer compatible both with the IBM network just described and with the Eclipse installations at RTI and the collaborating medical centers; (3) a Summagraphics MM1200 bit pad for digitizing cochlear cross sections for input to the field-neuron model calculations; (4) a Hewlett-Packard multi-pen plotter for documenting field-neuron model results; and (5) an additional graphics terminal for use with the Eclipse S/140.

Added to NPO's existing Data General and IBM-PC computers, this equipment will complete a highly efficient research, software development, and word processing environment to support our professional staff both on site at RTI and when travelling to any of the collaborating medical centers.

C. Collaborative Arrangements

We have made collaborative arrangements for the conduct of evaluation studies at ~~the~~^{the} University of California at San Francisco (UCSF), Duke University Medical Center (DUMC), and ^{at} Washington University and Central Institute for the Deaf in St. Louis (Wash. U./CID). Tests with implant patients at these three centers will provide a large base of data for evaluating differences in the performance of speech processing strategies among subjects. Subcontracts and consulting arrangements are suggested for this contract to support the indicated efforts of the following individuals:

<u>name</u>	<u>responsibilities</u>
Mark White, Ph.D.	<ol style="list-style-type: none">1) Direct and coordinate collaborative effort in San Francisco.2) Help design and conduct tests on stimulus primitives, single channel coding strategies, and multichannel coding strategies.3) Collaborate in the further development and application of the integrated field-neuron model, to be used in this project to design speech processors with improved knowledge of the "electrical-to-neural transformer" that links outputs of the speech processor to inputs of the central nervous system.4) Provide advice on speech coding strategies.5) Provide advice and assistance on the use of existing UCSF hardware and software to help implement automated procedures for obtaining measures of basic psychophysical performance and speech understanding.6) Help design and conduct intracochlear evoked potential tests to evaluate neural

survival patterns, neural physiological integrity and ["]electrical-to-neural transformer" characteristics of implanted patients.

Margaret Skinner, Ph.D.

- 1) Direct and coordinate collaborative effort in St. Louis.
- 2) Help design and conduct tests on stimulus primitives, single channel coding strategies and multichannel coding strategies.
- 3) Provide advice on the development of testing procedures
- 4) Provide advice on speech coding strategies.

Maynard Engebretson, Ph.D.

- 1) Help establish and maintain the Eclipse lab in St. Louis.
- 2) Provide advice on development of TMS320-based hardware systems.
- 3) Provide advice on speech coding strategies.

Bruce Weber, Ph.D.

- 1) Direct and coordinate collaborative effort at Duke.
- 2) Help design and conduct tests on stimulus primitives, single channel coding strategies, and multichannel coding strategies.
- 3) Provide advice on the development of testing procedures and on the measurement and interpretation of intracochlear evoked potentials.
- 4) Provide advice on the development of testing procedures.

In addition to the collaborations indicated above, we have a collaborative agreement with Storz Instrument Company of St. Louis to evaluate single channel coding strategies. Our ongoing collaborations with

UCSF, DUMC, and Storz are fully described in section II.F of this proposal. Formal letters of collaboration from the groups at UCSF, Wash. U./CID, and DUMC are reproduced on the following pages.

**John C. & Edw. Coleman Memorial Laboratory
Saul and Ida Epstein Otoneurological Laboratory**

871 HSE

University of California at San Francisco

San Francisco, CA 94143

Telephone (415) 666-2511

April 29, 1985

Dr. Blake S. Wilson
Neuroscience Program Office
Research Triangle Institute
Research Triangle Park, NC 27709

Dear Blake,

This letter is written to acknowledge our continued enthusiasm for collaborating with you on your development of speech processors for cochlear prostheses, supported by the Neuroprosthesis Contract Program. We believe that progress in your initial contract period has been outstanding. Your construction of the Block Diagram Compiler speech processor model is of tremendous importance to our group. With the beautiful interface hardware that you have provided, we'll make powerful use of this instrument (with your collaboration) in the next UCSF experimental patient (to be implanted on May 8). Your progress in e-field modeling of intracochlear electrodes has also been outstanding, and highly relevant to your -- and our -- long range objective of efficiently defining what you have termed "stimulus primitives". We (especially Mark White) are making good use of these data already. I believe that with Mark White's psychophysical studies, a quantum leap has been achieved in the understanding of basic electrophysiological considerations underlying control of an intracochlear electrode array.

To further support the RTI-USCF collaboration, we are recruiting a speech psychophysicist to work with your speech processing model at UCSF. The search committee has now selected a candidate for this position. This young Ph. D., Dr. John Kingston, is thoroughly familiar with Eclipse systems and with ILS software, and should be of immediate help. If he accepts this position, he will join the group around June 1. With Mark White, a doctoral student (David Morledge) and our engineering personnel collaborating in these experiments, we believe that we can strongly support the evaluation of your model at UCSF, and will continue to use it heavily over your proposed renewal period.

As you know, we are also delighted to hear of your possible collaboration with the excellent research group at CID and the Otolaryngology Department at Washington University. This is another of the several research teams really capable of making substantial progress in prosthesis speech processor development. We shall support the Washington University group in whatever way we can from our end.

We are absolutely delighted to continue our collaboration with you, and, indeed, look forward to even stronger intergroup ties. Mark White's possible direct work with you on

consideration of electrode-nerve interface studies is a case in point. As you appreciate, Mark is the world's authority in this area re auditory nerve stimulation. This collaboration should be a highly fruitful one for all concerned.

Keep up the good work. We look forward to Charlie and you joining us in June for another intensive research period. Let's hope that this patient is one with good nerve survival!

Yours,

Michael M. Merzenich
Director, Coleman Laboratory
Professor, Physiology & Otolaryngol.

WASHINGTON
UNIVERSITY
SCHOOL OF
MEDICINE
AT WASHINGTON UNIVERSITY MEDICAL CENTER

DEPARTMENT OF OTOLARYNGOLOGY-
HEAD AND NECK SURGERY

Division of Audiology

May 2, 1985

Blake S. Wilson, Ph.D., Manager
Neuroscience Program
Research Triangle Institute
Post Office Box 12194
Research Triangle Park, North Carolina 27709

Dear Blake:

This letter is to confirm our commitment to collaborate with your group, the team at the University of California - San Francisco Medical Center and the team at Duke University Medical Center as part of your contract proposal to NIH-NINCDS (Number 85-09) on "Speech Processors for Auditory Prostheses" that you are submitting in May 1985.

We feel it is an honor and privilege to be asked to collaborate with you. The expertise which you, Charles Finley, Dewey Lawson, Michael Merzenich and all his collaborators at UCSF have gained through extensive research with cochlear implants ranks among the best in the world today. In our opinion, your group, the group at UCSF and the group at Duke are asking the most important research question, that is, "What signal processing scheme(s) will provide the most benefit to patients who receive cochlear implants?" Your groups have developed the powerful tools (an optimized multichannel electrode array, percutaneous cable, computer/patient interface, computer-based simulator for implementation of many processors in tests with single subjects, computer-controlled stimulation at single or multiple pairs of electrodes to evaluate neuron survival in single subjects, and an integrated field-neuron model of intracochlear electrical stimulation) that make it possible to collect relevant data for answering this question now. Furthermore, we believe your emphasis on "defining the characteristics of the 'electrical-to-neural transformer' in the implanted ear" is crucial to designing speech processors which may prove more fruitful than focusing solely on coding the acoustic properties of speech in your search for the best processor.

We believe that we can make a significant contribution to your project. We have already formed a cochlear implant team including ourselves, Peter G. Smith, M.D., Ph.D., Susan M. Binzer, M.A. (Coordinator and aural rehabilitationist), and Elizabeth J. Nettles, Ph.D. (Psychologist), and we have been evaluating patients since September 1984. We do have access to patients from a fairly wide geographical area, and Washington University

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WASHINGTON UNIVERSITY SCHOOL OF MEDICINE

Blake S. Wilson, Ph.D.

Page Two

School of Medicine has an excellent Public Relations Department that has already helped us and will in the future. Our facilities for 1) the preoperative evaluation of patients both medically, audiological, and psychologically, 2) the surgical implantation and hospitalization, 3) the experimental evaluation of patients, 4) the aural rehabilitation and counselling of patients and their families, and 4) the long-term care of patients are excellent. Furthermore, we have the resources to make a contribution to research which will hopefully lead to better cochlear implants. Our Audiological Research Laboratory will be finished in June 1985, and we have a three-year grant from the Edward Mallinckrodt, Jr. Foundation for salary support for the evaluation of cochlear implant patients. Barbara A. Bohne, Ph.D., who is an anatomist of the inner ear with our department, is arranging to study the relation of dimensions of chinchilla and human ears measured anatomically to those measured on high-resolution CT scans with Mohktar Gado, M.D., who is chief of neuroradiology. In addition, we have a close working relationship with Maynard Engebretson, Ph.D. and his colleagues at Central Institute for the Deaf and Washington University, who are very interested in the development of wearable digital speech processors.

We will obtain a computer system which duplicates that at RTI and Duke including an Eclipse S-140 with floating point instruction set, a minimum of 512 Kb memory, a DCU-50 digital control unit, a backplane-paddleboard cable for the DCU-50, a bus extender cable, a backplane for DCU-50 I/O devices, a 4066 digital I/O card, a 9-track, 75 ips, 800 ppi mag tape subsystem, including a 6021 controller card, a winchester disk mass storage device (e.g. 6103 [25Mb] or 6099 [12.5 Mb], a cabinet, a console CRT (e.g., D-200), and a second asynchronous port for communications (e.g., AMI-8 board). Maynard Engebretson has agreed to guide us in assembling this equipment. It is our understanding that you will build the computer/patient interface for us, hopefully as part of the NIH contract.

We agree to make an intensive effort to find and evaluate appropriate experimental subjects, arrange for funding of their implants, and follow the experimental protocol that you design, with the understanding that we can add a test or two of particular interest to us. We also agree to keep records of all the data we collect and send these to you at mutually agreed upon times; the confidentiality of these records will be maintained according to the guidelines of the Internal Review Board at Washington University Medical School. We will need guidance from you in implementing the protocols for the experimental subjects, and we will want to communicate with you, and the groups at Duke, UCSF and Storz Instrument Company often. We also agree for Margo to devote 10% time and effort to development of speech processing models on your contract proposal, which you discussed on April 22; the attached budget page shows the amount.

We are enclosing a brief summary of the facilities and a floor plan of these, and a copy of our curriculum vitae. A copy of the signed agreement with Storz Instrument Company and a copy of the application we have made to our Internal Review Board will be sent as soon as IRB approval is granted.

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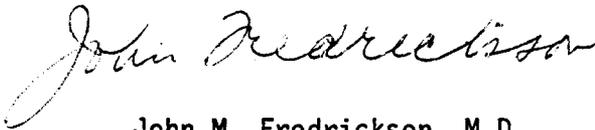
Blake S. Wilson, Ph.D.
Page Three

Although we have not received approval yet, we do not anticipate any difficulty since we have had approval for implanting the 3M/Vienna extracochlear implant for over a year.

We look forward with excitement and pleasure to our collaboration with you.

Please let us know if you need further information from us.

Sincerely,



John M. Fredrickson, M.D.
Lindburg Professor and Head



Peter G. Smith, M.D., Ph.D.
Assistant Professor

cc: Maynard Engebretson, Ph.D.
Michael M. Merzenich, Ph.D.
David Calvert

enclosures



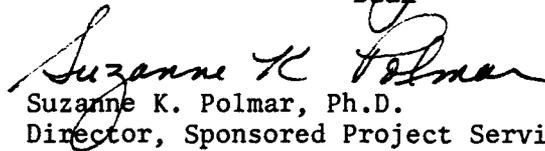
Margaret W. Skinner, Ph.D.
Director of Audiological Services



Susan M. Binzer, M.A., CCC-A
Coordinator, Cochlear Implant Program



M. Kenton King, M.D.
Dean



Suzanne K. Polmar, Ph.D.
Director, Sponsored Project Services

DESCRIPTION OF FACILITIES IN THE DEPARTMENT OF OTOLARYNGOLOGY,
WASHINGTON UNIVERSITY SCHOOL OF MEDICINE

The resources available to conduct the experimental protocol for the contract "Speech Processors for Auditory Prostheses" are excellent, in terms of access to clinical populations, medical, audiological and psychological evaluation of prospective candidates, surgical implantation, hospitalization, post-surgical experimental evaluation, aural rehabilitation and counseling of patients and their families, and long-term care.

Each prospective candidate will be examined by either John M. Fredrickson, M.D. or Peter G. Smith, M.D., Ph.D., their vestibular function will be evaluated in our recently updated Neuro-otology Laboratory, and their audiological function will be evaluated in our Division of Audiology. These evaluations will all be done in our clinical facilities on the 8th Floor of McMillan Hospital (see the attached floor plan). Susan M. Binzer, M.A., CCCA, will evaluate each patient's communication skills including speechreading and counsel them about cochlear implants. This evaluation will be done in the Audiological Research Laboratory (see attached floor plan) which consists of a therapy room, a double-walled sound booth, and a room containing the computers, sound generating and video equipment, and equipment for collecting, analyzing and printing out the results. There is also equipment for the calibration of the stimuli. The radiological evaluation will be done at Mallinckrodt Institute which is part of the Washington University Medical Center; the results will be evaluated jointly with Mokhtar Gado, M.D. and Barbara A. Bohne, Ph.D. The psychological evaluation will be done in our department by Elizabeth J. Nettles, Ph.D., who is in private practice and has had a number of years experience evaluating and counseling deaf people. She will be available if a patient needs further counseling.

The surgical implantation will be done in recently renovated, state-of-the-art operating rooms in Barnes Hospital, which serves as the primary teaching affiliate for the Washington University School of Medicine. Hospitalization following surgery will be in Barnes Hospital on the floor devoted to patients from the Department of Otolaryngology.

The experimental evaluation and aural rehabilitation of patients will be done in the Auditory Research Laboratory. The long-term care will be given in this laboratory and in the clinical facilities on the 8th floor of McMillan Hospital.

A. Maynard Engebretson, Ph.D., who is Assistant Director of Research in Engineering at Central Institute for the Deaf and Associate Affiliate Professor in the Department of Computer Science Department as well as Associate Professor of Electrical Engineering, Department of Speech and Hearing of Washington University, has agreed to consult with us on the computer system.

David Calvert, who is Director of the Cochlear Implant Project for Storz Instrument Company, has agreed to work closely with us as we participate in the experimental protocol. In addition, he has agreed to loan us the equipment for testing the implant in the operating room and for adjusting the clinical device processor when the experimental protocol is completed. Storz is located in St. Louis which will facilitate this collaboration.

CENTRAL INSTITUTE FOR THE DEAF

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PROFESSIONAL TRAINING
HEARING, LANGUAGE, SPEECH CLINICS
PARENT-CHILD LANGUAGE TRAINING PROGRAMS

May 6, 1985

Blake S. Wilson, Ph.D.
Head, Neurosciences Program
Research Triangle Institute
Research Triangle Park, NC 27709

Dear Blake:

I would be happy to participate in your project entitled "Speech Processors for Auditory Prostheses". I can be available at a 5% annual effort to consult with you on real-time signal processing systems and to work with Dr. Margaret Skinner to help install and to make operational an S140 computer system for evaluating cochlear implant models.

Sincerely yours,

A. Maynard Engebretson, D.Sc.
Assistant Director of
Research in Engineering

AME/db

Duke University Medical Center

DURHAM NORTH CAROLINA 27710

DEPARTMENT OF SURGERY
DIVISION OF OTOLARYNGOLOGY
P. O. BOX 3805

OFFICE (919) 684-6968
APPOINTMENTS 684-3834

May 14, 1985

Mr. Blake S. Wilson, Director
Neuroscience Program Office
Research Triangle Institute
Research Triangle Park, North Carolina 27709

Dear Blake:

This letter is to indicate our enthusiasm for our continued collaboration with you on your projects for the development of speech processors for cochlear prostheses. We have been very impressed with the progress that you have made with your first contract for this project, and your work has been very influential in our efforts to begin a cochlear implant program at Duke University Medical Center. In response to your input to our program, Dr. David C. Sabiston, Jr., Chairman of the Department of Surgery, has authorized monies to assist in the construction of the interface hardware that you designed for testing patients with your Computer-based Simulator of Speech Processors. Your interface hardware and your computer simulation techniques will be essential components in our clinical implant program.

As you know, we are actively assessing applicants for a cochlear implant, and once a suitable candidate is implanted, we will use your computer-based methods to determine the signal processing schemes that are best for that patient. All the members of our team believe that your approach--that of assessing many signal processing schemes--will provide the most benefit to the cochlear implant patient.

During the course of assessing candidates for cochlear implants we shall make every effort to identify those patients who would be suitable experimental subjects, and we will actively seek funding for the implants of these patients. Such experimental patients will undergo tests described in your experimental protocol and in your contract applications to NIH as well as in other written material to us.

We think we have made a useful contribution to your project in the past, and we intend to contribute in the future. We have an active research program in the anatomy and physiology of the central auditory system. Members of this program are willing to assist you in any way they can. We would like to add that the long term goal of our research program is to attract additional scientists who would interface between our basic research program and the clinical application of cochlear implants.

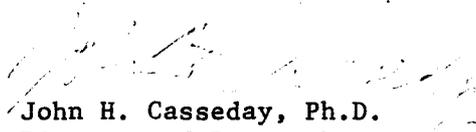
Mr. Wilson
Page 2
May 14, 1985

We look forward to the continuation of an extremely productive collaboration with you.

Sincerely,



William R. Hudson, M.D.
Professor and Chief



John H. Casseday, Ph.D.
Director of Research
Associate Medical Research Prof.

WRH/ts

Duke University Medical Center

DURHAM, NORTH CAROLINA
27710

DEPARTMENT OF SURGERY
CENTER FOR SPEECH AND
HEARING DISORDERS

TELEPHONE (919) 684-3859
P. O. BOX 3887

May 10, 1985

Blake S. Wilson, Head
Neuroscience Program Office
Research Triangle Institute
Research Triangle Park, NC 27709

Dear Blake,

I am writing to provide you with a formal statement of the enthusiastic support of the Speech Pathology and Audiology Programs for your development of speech processors for cochlear prostheses. We are delighted to have an opportunity to collaborate with you in this endeavor.

We believe that the close proximity of Duke Medical Center and RTI would allow us to work closely in providing a superior cochlear implant system for the profoundly hearing impaired. The Center for Speech and Hearing Disorders at Duke, as part of a major medical center, is providing ongoing clinical services to a large number of hearing impaired individuals. This should serve as an excellent source for cochlear implant candidates who would agree to serve as research subjects in your project. The newly established multi-disciplinary Center for the Severely Hearing Impaired at Duke should further enhance our ability to identify individuals who can profit from the UCSF - Storz cochlear implant device. We will make every effort to ensure that these patients are aware of the importance of your research and your need for research subjects.

I am personally pleased to be involved in your research efforts and will assume responsibility for coordinating the activities here at Duke. I will also provide assistance in the development of testing strategies and protocols.

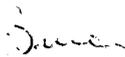
The entire staff here at the Center for Speech and Hearing Disorders will cooperate with you in the conduct of your research activities. As you are aware, we have allocated space for the testing of implant candidates and have arranged for renovations to provide suitable housing for the Eclipse computer. I believe that Duke financial assistance in the duplication of the RTI computer here in the Center also attests to strong support for your project.

Blake S. Wilson
May 10, 1985

-2-

We are most pleased that there is an opportunity for us to become more closely involved with your research endeavors. We strongly believe that these efforts will result in a speech processor which will be of great benefit to the profoundly hearing impaired. We are delighted to work with you toward this most important goal.

Sincerely,



Bruce A. Weber, Ph.D.
Associate Professor

BAW:mf

D. Resumes

BLAKE S. WILSON

Experience

1984 to date: Adjunct Assistant Professor of Experimental Otolaryngology, Duke University Medical Center (DUMC). The primary responsibility for this position is to direct research to develop and evaluate auditory prostheses implanted in patients at DUMC. Additional responsibilities are to (1) coordinate research efforts at DUMC with those at the University of California at San Francisco (UCSF); (2) assist in the development of procedures used in the clinic for evaluation and treatment of cochlear-implant patients; and (3) participate in other activities of DUMC's "Center for the Severely Hearing Impaired," one of which is the development and clinical application of advanced, signal-processing hearing aids.

1974 to date: Several positions at the Research Triangle Institute (RTI), including Research Engineer ('74-'78); Senior Research Engineer ('78-date); and Head, Neuroscience Program Office ('83-date). Experience in these positions is indicated below and on the attached list of projects for which B. Wilson was or is the principal investigator.

Principal Investigator of a project to develop speech processors for multichannel auditory prostheses. Major tasks in this project include the following: (1) identify and contrast the most promising approaches to the design of speech processors for multichannel prostheses; (2) build a computer-based simulator that is capable of rapid and practical emulation of all these approaches in software; (3) design and fabricate a hardware interface that will provide a communications link between the computer and implanted electrodes; and (4) evaluate promising strategies for speech processing in tests with single subjects so that meaningful comparisons of performance can be made. (Funded by the NINCDS, 1983-85, \$398k)

Principal Investigator of a feasibility study to evaluate the efficacy of single-channel coding strategies for extra-cochlear auditory prostheses. The computer-based simulator mentioned above will be used in this study to evaluate four coding strategies for single-channel auditory prostheses. (Funded by Storz Instrument Company, 1984-86, \$160k)

Principal Investigator of a study to identify the sites at which nonionizing radiation acts in eliciting auditory responses. Methods that were used include: (1) recording and analysis of responses of single units in the auditory nerve and cochlear nucleus to pulses of microwave irradiation and (2) mapping of activity-related alterations in brain metabolism induced by exposures to continuous-wave and pulsed microwave radiation. Patterns of brain metabolism were measured by autoradiographic survey of [14 C]-2-deoxy-D-glucose uptake in neural tissue. (Funded by the NIEHS, 1979-81, \$94k)

Principal Investigator of a study to evaluate possible effects of nonionizing radiation on brain structures outside of the auditory system. Detection of radiation-induced alterations in the activities of the vestibular system, hypothalamus, and cerebral cortex was emphasized. Patterns of brain activity were again measured using the [14 C]-2-deoxy-D-glucose method. (Funded by the EPA, 1979-83, \$157k)

Consultant to (1) the Veterans Administration and the NSF for site visits and reviews of proposals in the areas of microwave bioeffects and speech and hearing prostheses (1979-80); (2) the University of California at San Francisco for the design of speech processors for auditory prostheses (1978); and (3) the NIEHS for the designs of an auditory-nerve simulator and a distortion-compensating driver for B&K transducers (1980-82).

Project Leader at RTI of a collaborative study to improve the mechanical design of vascular prostheses used for small-artery replacement. To explore the relationships between mechanical design parameters and velocities of flow, stresses at the sutures, and coefficients of reflected energy at the proximal anastomosis, we developed a computer simulation model of blood flow dynamics in the healthy leg and in legs in which a femoro-popliteal bypass graft had been inserted. This model was based on finite-difference solutions of the one-dimensional, Navier-Stokes equations of fluid flow in compliant tubes. (Funded by the VA, 1978-79)

Principal Investigator of a study to evaluate the [¹⁴C]-2-deoxy-D-glucose method for use in research on the biological effects of microwave radiation. (Funded by the NIEHS, 1976-77)

Project Leader for software development of microprocessor-based, six station audiometer. (Funded by Monitor, Inc., 1976-77)

Principal Investigator of a study to record and compare responses in the auditory nerve to acoustic clicks and to pulses of microwave irradiation. (Funded by the EPA, 1975)

Engineer in a study to develop a speech-analyzing lipreading aid for the profoundly deaf based on the cued-speech system of visual speech communication. Speech analysis work included the design of a voice pitch extractor, development of advanced algorithms for speech segmentation, and computer recognition of speech phonemes in real time. Other work on this project included computer simulation of the "Automatic Cued" output for training and testing of deaf subjects. (Funded by the NINCDS and Gallaudet College, 1974-76, and by NASA and the VA thereafter; B. Wilson's participation was in the years 1974-76)

Member of a biomedical applications team to define unsolved problems in biomedical technology through consultation with research physicians and then to identify solutions for these problems with the aid of NASA technology. (Funded by NASA; B. Wilson's participation was in the years 1974-75)

1971 to 1973: Electronics technician at the Duke University Medical Center, Department of Surgery.

Responsibilities for this position were design and fabrication of instruments for biomedical research; design and fabrication of equipment to monitor and ensure patient safety in the operating rooms; repair of critical electronic equipment in the operating rooms; provide instruction on the use of physiological monitoring systems for operating room personnel.

Education

B.S., Electrical Engineering, Duke University, 1974.

Professional Honors and Activities

Recipient of Professional Development Awards from the Research Triangle Institute to (1) develop speech processors for auditory prostheses (1977) and (2) participate in an expedition sponsored by the National Geographic Society to elucidate the acoustic basis of prey recognition by mustache bats (1983).

Appointment to the subcommittee on microwave and laser exposure of the N.C. Radiation Protection Commission, 1981-date.

Discussion Leader, Gordon Research Conference on Implantable Auditory Prostheses, August, 1985.

Chairman of the session on Cardiovascular Fluid Dynamics, 2nd Mid-Atlantic Conference on Bio-Fluid Mechanics, April, 1980.

Member of the IEEE (special section on Engineering in Medicine and Biology), Society for Neuroscience (N.C. Chapter), and the Bioelectromagnetics Society.

Selected Papers and Abstracts

Wilson, BS: Mechanisms and physiologic significance of microwave action on the auditory system. J. Bioelectricity (invited paper in press).

Kobler, JB, BS Wilson, OW Henson, Jr. and AL Bishop: Echo intensity compensation by echolocating bats. Hearing Res. (submitted).

Wilson, BS and CC Finley: Speech processors for auditory prostheses. To be presented at the International Cochlear Implant Symposium and Workshop, Melbourne, Australia, Aug. 27-31, 1985 (full-length paper to be published in the proceedings).

Finley, CC and BS Wilson: Field models of the Melbourne electrode array. Invited paper to be presented at the International Cochlear Implant Symposium and Workshop, Melbourne, Australia, Aug. 27-31, 1985 (full-length paper to be published in the proceedings).

Wilson, BS: Coding strategies for multichannel auditory prostheses. Invited paper to be presented at the 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 to be 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 to be presented in the special session on signal processing for the hearing impaired, IEEE Bioengineering Conf., Sept. 27-30, 1985 (full-length paper to be published in the proceedings).

Wilson, BS and CC Finley: A computer-based simulator of speech processors for auditory prostheses. ARO Abstracts, 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. ARO Abstracts, 8th Midwinter Research Conference, p. 105, 1985.

Henson, OW, Jr., BS Wilson, JB Kobler, AL Bishop, MM Henson and R Hansen: Radiotelemetry and computer analysis of biosonar signals emitted by the free-flying bat. ARO Abstracts, 8th Midwinter Research Conference, p. 70, 1985.

- Henson, OW, Jr., JB Kobler, BS Wilson and AL Bishop: Echo frequency and intensity optimization by mustache bats. ARO Abstracts, 7th Midwinter Research Conference, 1984.
- Wilson, BS, JB Kobler, JH Casseday and WT Joines: Spectral content of microwave-induced auditory stimuli as demonstrated by [^{14}C]-2-deoxy-D-glucose uptake at the inferior colliculus. Bioelectromagnetics Abstracts, 5th Annual Meeting, p. 46, 1983.
- Blackman, CF and BS Wilson: Distribution of label in studies on the effects of nonionizing radiation on the association of calcium ions with brain tissue. Bioelectromagnetics Abstracts, 5th Annual Meeting, p. 73, 1983.
- Wilson, BS, JM Zook, WT Joines and JH Casseday: Alterations in activity at auditory nuclei of the rat induced by exposure to microwave radiation: Autoradiographic evidence using [^{14}C]-2-deoxy-D-glucose. Brain Res., 187: 291-306, 1980.
- Joines, WT and BS Wilson: Field-induced forces at dielectric interfaces as a possible mechanism of rf hearing effects. Bull. Math. Biol., 43: 401-413, 1981.
- Wilson, BS, WT Joines, JH Casseday and JB Kobler: Responses in the auditory nerve to pulsed, CW, and sinusoidally-modulated microwave radiation. Bioelectromagnetics, 1: 237, 1980.
- Wilson, BS, WT Joines, JH Casseday and JB Kobler: Identification of sites in brain tissue affected by nonionizing radiation. Bioelectromagnetics, 1: 208, 1980.
- Scott, SM and BS Wilson: The mechanical design of vascular prostheses. In P Puel, H Boccalon and A Enjalbert (Eds.), Hemodynamics of the Limbs, Institut National de la Sante et de la Recherche Medicale, Paris, 1979, pp. 251-259.
- Wilson, BS and SM Scott: Hemodynamic design considerations for an improved artery shunt prosthesis. In DJ Schneck (Ed.), Bio-Fluid Mechanics, VPI Press, Blacksburg, Va., 1978, pp. 93-98.
- Cornett, RO, RL Beadles and BS Wilson: Automatic cued speech. In JM Pickett (Ed.), Papers from the Research Conference on Speech-Processing Aids for the Deaf, May 24-26, 1977, published by Gallaudet Research Institute, Gallaudet College, Washington, D.C., 1981, pp. 224-239.
- Wilson, BS: Problems and opportunities in the design of speech processors for cochlear implant prostheses. Proc. AAMI Annual Meeting, 13: 295, 1978.
- Joines, WT, BS Wilson and S Sharp: Temperature-controlled heating of tumors by microwaves. Proc. of the IEEE Southeastern Conference, 1977, pp. 124-128.
- Wright, D, J Hebrank and B Wilson: Pinna reflections as cues for localization. J. Acoust. Soc. Am., 56: 957-962, 1974.

Major Reports

- Wilson, BS: Quarterly reports for NIH projects N01-ES-9-008 and N01-NS-2356 and for EPA projects 68-02-2231 and 68-02-3276. (see last page for project titles and dates)
- Wilson, BS: Identification of sites in brain tissue affected by nonionizing electromagnetic radiation. Final report, EPA project 68-02-3276, November, 1983.
- Wilson, BS: Critical review of methods used for diagnosing depth of injury in burn victims. May, 1982. (supported by a NASA contract to D Rouse, "Biomedical Applications Team")
- Wilson, BS, JH Casseday and JB Kobler: Laboratory computer facility for auditory research. August, 1981. (supported by NIH contract N01-ES-9-008)
- Wilson, BS and JB Kobler: Cochlear nerve simulator. Final Report, NIH project PR-048281, February, 1981.
- Wilson, BS: Investigations to determine the peripheral and central receptors mediating effects of microwave radiation on brain activity. Final report, NIH project N01-ES-9-008, August, 1981.
- Wilson, BS and RL Beadles: Real time speech analysis and display software for the automatic cuer. Final report for Gallaudet College contract 37301, November, 1976.
- Wilson, BS and RL Beadles: A pitch extractor for real-time applications of speech analysis. 1975. (supported by an NIH contract to RL Beadles, "Develop a Speech Autocuer")

Invited Paper in Preparation

- Kenan, PD, BS Wilson, BA Weber and JC Farmer, Jr.: Center for the severely hearing impaired. (to be published in NC Medical J., 1985)

Other Manuscripts in Preparation

- Wilson, BS and CC Finley: Speech processors for auditory prostheses. Part I. Coding strategies. (to be submitted for publication in Hearing Res.)
- Wilson, BS and CC Finley: A computer-based simulator of speech processors for auditory prostheses and advanced, signal-processing hearing aids. (to be submitted for publication in Hearing Res.)
- Wilson, BS, JH Casseday, JB Kobler and WT Joines: Auditory responses to microwave radiation. Part I. Single-unit studies. (to be submitted for publication in J. Acoust. Soc. Am.)
- Wilson, BS, JB Kobler, JH Casseday and WT Joines: Auditory responses to microwave radiation. Part II. Modeling studies. (to be submitted for publication in J. Acoust. Soc. Am.)
- Wilson, BS, JB Kobler, JH Casseday and WT Joines: Effects of microwave radiation on brain activity. (to be submitted for publication in Brain Res.)

Inventions

Patent disclosures have been written to describe (1) three-dimensional reproduction of sound using two channels of information; (2) circuits to enhance the intelligibility of speech for hearing aid users; (3) a system for improved reproduction of sound in headphones; (4) a system for reduction of noise mixed with certain quasi-periodic signals; (5) a system for improved reproduction of perceived loudness in stereo playback units; and (6) a system for electronic control of spark advance in gasoline engines.

Patent disclosures that describe 13 inventions in the general area of cochlear implants are in preparation.

Chronological Record of Projects Directed by Blake Wilson

<u>Period</u>	<u>Agency & Number</u>	<u>Title</u>	<u>Amount</u>
'75	EPA 68-02-2231	Auditory Neural Response to Low-Power Pulsed Microwave Irradiation: A Pilot Study	\$18,985
'76-'77	NIH PD-151468-7	Evaluate the Efficacy of [¹⁴ C]-2-deoxy-D-glucose Autoradiography for Mapping Brain Responses to Microwave Radiation	10,000
'76-'77	Monitor, Inc.	Computerized Audiometer Software	17,751
'77	RTI (PDA)	Speech Encoders for Cochlear Implant Prostheses	5,866
'78-'79	VA, various #'s	Develop Computer Models of Blood Flow Dynamics in the Healthy Human Leg and in Legs in Which Femoro-Popliteal Bypass Grafts Have Been Inserted	13,800
'79-'81	NIH N01-ES-9-008	Investigations to Determine the Peripheral and Central Receptors Mediating Effects of Microwave Radiation on Brain Activity	94,268
'79-'83	EPA 68-02-3276	Identification of Sites in Brain Tissue Affected by Nonionizing Electromagnetic Radiation	156,681
'81	NIH PR-048281	Auditory Nerve Simulator	4,679
'83	RTI (PDA)	Participate in an Expedition Sponsored by the National Geographic Society to Elucidate the Acoustic Basis of Prey Recognition by Mustache Bats	10,284
'83-'85	NIH N01-NS-2356	Speech Processors for Auditory Prostheses	397,926
'84-'86	Storz Inst. Co.	Evaluate the Efficacy of Single-Channel Coding Strategies for Extra-Cochlear Auditory Prostheses	160,000

NAME: Finley, Charles Coleman SS No: 420-66-4216

DATE OF BIRTH: January 10, 1949

PLACE OF BIRTH: Florence, Alabama CITIZENSHIP: U.S.A.

MAJOR INTEREST: Neurophysiology of Sensory Mechanisms

EDUCATION:

Georgia Institute of Technology, Atlanta, Georgia
BS in Electrical Engineering/Co-operative Program 67-72

Swiss Federal Technical Institute (ETH), Zurich, Switzerland
Electrical Engineering Exchange Student 72-73

University of North Carolina at Chapel Hill
PhD in Neurobiology 73-83

EXPERIENCE:

Project Director of Ga. Tech Electric Vehicle Design Program 70-72

Co-op Student - Electronic Design Laboratory, Central Intelligence
Agency, Washington, D.C. 68-71

Research Assistant - Department of Neurology, Kanton Hospital,
Zurich, Switzerland 72-73

Electronic Technician - Medical School Electronics Laboratory
University of N.C. at Chapel Hill 81-83

Private Consultant in development of multiple-subject, computer-
based audiometric testing system 76-83

Neuroscientist/Engineer - Neuroscience Program Office, Research
Triangle Institute 84-Present

HONORS:

Tau Beta Pi Phi Kappa Phi Eta Kappa Nu

Selected Outstanding Student Engineer 1972 by the Professional
Engineers of Greater Atlanta

Recipient of the World Student Fund Scholarship of Ga. Tech 1972

Professional Development Award - Research Triangle Institute 1985

PUBLICATIONS:

- Henn V, Young LR and Finley C (1974): Vestibular nucleus units in alert monkeys are also influenced by moving visual fields. *Brain Research* 71: 144-149.
- Kaufmann P, Finley C, Bennett P and Farmer J (1979): Spinal cord seizures elicited by high pressures of helium. *Electroenceph. clin. Neurophysiology* 47: 31-40.
- Finley C (1983): Monitoring primary vestibular afferents during active, free-ranging behavior of the gerbil: Access to centrifugal control functions of efferent fibers. Dissertation, University of North Carolina at Chapel Hill.
- Finley C, Frye G and Breese G (1985): Brainstem evoked potential correlates of neural supersensitivity in rats following propylthiouracil treatment. (in preparation)
- Dawson G, Finley C, Phillips S and Galpert L (1985): Hemispheric specialization and language acquisition in autistic children. (submitted).
- Dawson G, Finley C, Phillips S and Galpert L (1985): Cognitive processing of verbal and non-verbal stimuli in autistic children as indexed by the P300 component of the auditory evoked potential. (submitted).

ABSTRACTS, POSTERS AND PRESENTATIONS:

- Henn V, Young L and Finley C (1973): Vestibular nucleus units in alert monkeys are also influenced by moving visual fields. *Abstr. in Fortschritte der Zoologie* 23(1): 247.
- Henn V, Young L and Finley C (1974): Unit recording from the vestibular nucleus in the alert monkey. *Abstr. for Workshop Meeting of the European Brain and Behavior Society - Vestibular Functions and Behavior, April 25, Pavia (Italy)*
- Finley C, Bennett P, Farmer and Kaufmann (1978): High pressure nervous syndrome at the spinal level in rats. Presentation and abstr. for the Undersea Biomedical Society, *Undersea Biomedical Research* 5: 44.
- Finley C (1983): Expanded capabilities of audiometric testing using microprocessors. Presentation for N.C. Acoustical Society of America, April 28, Spring Meeting.

- Martinkosky SJ, Howard JF, Finley CC and Quint SR (1983): Pre- and postplasmapheresis treatment measures obtained from myasthenia patients. Poster for American Speech and Hearing Association Meeting, Cinn., Ohio.
- Dawson G, Finley C and Frei T (1984): Hemisphere processing in echolalic and nonecholalic autistic children. Presented at the 1984 Meeting of the Body for Advancement of Brain, Behavior, and Language Enterprises (BABBLE), Niagra Falls, Ontario.
- Finley CC and Wilson BS (1985): An integrated field-neuron model of electrical stimulation by intracochlear scala-tympani electrodes. Abstr. Midwinter ARO Meeting.
- Wilson BS and Finley CC (1985): A computer-based simulator of speech processors for auditory prostheses. Abstr. Midwinter ARO Meeting.
- Finley CC (1985): Co-chairman for session on Cochlear Prosthetic Devices, 8th Annual ARO Meeting, Feb. 3-7, 1985.
- Dawson G, Finley C, Phillips S and Galpert L (1985): Hemispheric specialization and language development of autistic children. Accepted for presentation at the 1985 Meeting of the Society for Research in Child Development, Toronto, Canada.
- Finley CC and Wilson BS (1985): Models of neural stimulation for electrically evoked hearing. Invited presentation for the special session on neurostimulation, ACEMB Meeting, Sept. 30 - Oct. 2, 1985.
- Wilson BS and Finley CC (1985): Speech processors for auditory prostheses. Invited presentation for the special session on signal processing for the hearing impaired, IEEE Bioengineering Conference, Sept. 27-30, 1985 (full-length paper to be published in the proceedings).
- Finley CC (1985): An integrated field-neuron model of intracochlear stimulation. Invited presentation, Gordon Research Conference on Implantable Auditory Prostheses, Aug. 19-23, 1985.
- Wilson BS and Finley CC (1985): Speech processors for auditory prostheses. To be presented at the International Cochlear Implant Symposium and Workshop, Melbourne, Australia, Aug. 27-31, 1985 (full-length paper to be published in the proceedings).
- Finley CC and Wilson BS (1985): Field models of the Melbourne electrode array. Invited paper to be presented at the International Cochlear Implant Symposium and Workshop, Melbourne, Australia, Aug. 27-31, 1985 (full-length paper to be published in the proceedings).

Dewey Tall Lawson

Birthdate: February 6, 1944
Birthplace: Kinston, North Carolina, U.S.A.

Education: Smithfield, N.C., public schools
A. B. in Physics, Harvard University, 1966
Ph.D. in Physics, Duke University, 1972

Married: Elizabeth Booker Lawson, July 9, 1966
Children: Jonathan Dewey Lawson, born July 31, 1969
Neal Becton Lawson, born May 17, 1972

Professional Positions:

1985-present: Senior Scientist, Neuroscience Program Office, Research Triangle Institute, Research Triangle Park, N. C. Research and development related to speech processors for multichannel cochlear implants and single channel extracochlear prostheses; signal processing hearing aids.

1982-1984: Consultant in architectural and environmental acoustics and computer applications. Clients included architectural firms, professional associations, and private and governmental institutions. A principal client was Research Triangle Institute, continuing the projects mentioned below.

1979-1982: Senior Physicist, Center for Technology Applications, Research Triangle Institute, Research Triangle Park, N.C. Research on machine decoding of speech, ultrasonic and audible acoustics, hearing. Design and development of digital speech database, computer modeling of speech and speech decoding, computer modeling of ultrasonic experiments on flowing gas mixtures. Evaluation of research and development projects.

1980-present: Adjunct Associate Professor, Department of Physics, Duke University, Durham, N.C. Teaching "Acoustics and Music", a course offered jointly by the Physics and Music departments; occasional supervision of independent study students.

1974-1979: Assistant Professor and Research Associate, Department of Physics, Duke University, Durham, N.C. Teaching courses in physics, acoustics, and music. Supervision of graduate students in Low Temperature Physics. Research on collective mode excitations in superfluid helium-3. Developed an integrated system for control, data acquisition, analysis, and results presentation by a single computer.

Dewey Tull Lawson (continued)

1977-1978: Offered a Fulbright/Hays Senior Fellowship at the University of Sussex, but was unable to interrupt Duke research program at that time.

1972-1974: Research Associate, Laboratory of Atomic and Solid State Physics, Cornell University, Ithaca, N.Y. (postdoctoral fellowship). Research on collisionless ("zero") sound and hydrodynamics of superfluid helium-3. Developed a computer experimental control and data acquisition system.

1967-1971: Research Assistant and Associate, Department of Physics, Duke University, Durham, N.C. Research on heat transport and sound propagation in dielectric crystals at low temperatures. Developed an automated crystal growth apparatus and computer data analysis system.

Member:

American Physical Society
American Association of Physics Teachers
Acoustical Society of America
Sigma Xi

Publications: (see attached list)

University and Community Activities:

Former Chairman, Duke University Environmental Concerns Committee (composed of faculty, student, administrative, and community representatives; charged with advising the Chancellor on off-campus matters of concern to the University.)

First Presbyterian Church, Durham, N.C. Elder; deacon; member and assistant conductor of choir; chairman, Worship, Music, and Lecture Committee; teacher; former chairman, Task Force on Church and the Arts.

Former tympanist and principal of percussion, assistant conductor, chairman of the Orchestra Committee, and member of the board of directors, the Durham Symphony, Inc.

Former Durham YMCA children's soccer coach.

Boy Scouting: District committeeman, camporee chief, Order of the Arrow advisor, Cubmaster, Webelos Den Leader, troop committeeman, Scoutmaster; Eagle, Wood Badge, District Award of Merit, Scouter's Key.

Dewey Tull Lawson (continued)

Other Academic Interests:

The social, political, and art history of the city of Florence; especially art patronage during the republican administration of Piero Soderini (1502-1512).

Liturgics, in both historical and comparative aspects; especially the roles of music in liturgy.

Publication in Preparation:

"Computer Analysis of the Organ Works of Lambert Chaumont"

Scientific Publications of Dewey Tull Lawson

- "Phonon Scattering by Isolated Isotopic Lattice Perturbations in Single Crystals of Helium" (with H. A. Fairbank) Proc. of the 12th Internat. Conf. on Low Temp. Phys., Acad. Press of Japan, Tokyo, 1971, p.147 (abstract only).
- "Phonon Scattering by Isolated Isotopic Impurities in ⁴He Single Crystals" (with H. A. Fairbank) Bull. Am. Phys. Soc. 16, 638 (1971) (abstract only).
- "Phonon Scattering by Isolated Isotopic Impurities in Single Crystals of Helium" dissertation, Duke University, 1971.
- "Phonon Scattering by Isotopic Impurities in Helium Single Crystals" (with H. A. Fairbank) Low Temp. Phys. LT-13, Vol. II, Plenum, New York, 1973, p.85.
- "A Technique for Growing Helium Crystals in Preferred Orientations" Cryogenics 13, 276 (1973).
- "Thermal Conductivity and Isotopic Impurities in Single Crystals of Helium" (with H. A. Fairbank) J. Low Temp. Phys. 11, 363 (1973).
- "Sound Propagation through Liquid ³He in a Pomeranchuk Cell" (with W. J. Gully, S. Goldstein, R. C. Richardson, and D. M. Lee) Bull. Am. Phys. Soc. 18, 24 (1973) (abstract only).
- "Attenuation of Zero Sound and the Low Temperature Transitions in Liquid ³He" (with W. J. Gully, S. Goldstein, R. C. Richardson, and D. M. Lee) Phys. Rev. Lett. 30, 541 (1973).
- "The Effects of Magnetic Field on the 'A' Transition in Liquid ³He" (with W. J. Gully, D. D. Osheroff, R. C. Richardson, and D. M. Lee) Phys. Rev. A 8, 1633 (1973).
- "Magnetic Effects in Liquid ³He Below 3 mK" (with W. J. Gully, S. Goldstein, R. C. Richardson, and D. M. Lee) Bull. Am. Phys. Soc. 18, 642 (1973) (abstract only).
- "The Viscosity of Liquid ³He in the Fermi Liquid Region" (with W. J. Gully, S. Goldstein, J. D. Reppy, D. M. Lee, and R. C. Richardson) Bull. Am. Phys. Soc. 18, 642 (1973) (abstract only).
- "The Low Temperature Viscosity of Normal Liquid ³He" (with W. J. Gully, S. Goldstein, R. C. Richardson, J. D. Reppy, and D. M. Lee) J. Low Temp. Phys. 13, 503 (1973).
- "Attenuation of Zero Sound and the Several Low Temperature Phases of Liquid ³He" (with W. J. Gully, S. Goldstein, R. C. Richardson, and D. M. Lee) J. Low Temp. Phys. 15, 169 (1974).
- "Anisotropic Attenuation of Zero Sound in Superfluid ³He" (with H. M. Bozler and D. M. Lee) Bull. Am. Phys. Soc. 19, 1114 (1974) (abstract only).

- "Anisotropy in Superfluid ^3He and the Attenuation of Zero Sound"
(with H. M. Bozler and D. M. Lee) Phys. Rev. Lett. 34, 121 (1975).
- "Sound Propagation and Anisotropy in Liquid $^3\text{He-A}$ " (with H. M. Bozler and D. M. Lee) in Quantum Statistics and the Many-body Problem; S. B. Trickey, W. P. Kirk, and J. W. Dufty, editors; Plenum, New York, 1975, p.19.
- "Anisotropy and Sound Propagation in Superfluid ^3He " (with H. M. Bozler and D. M. Lee) Low Temp. Phys. LT-14, Vol. I, North-Holland, Amsterdam, 1975, p.92.
- "Superfluid Helium-3" in Yearbook of Science and Technology 1975; McGraw-Hill, New York, 1976, p.204.
- "Church Acoustics: Implications for Organ Performance", The Organ Yearbook, Vol. XI, p.116 (1980).
- "Interval-Based Representations of Complex Tones" Am. J. Phys. 48, 615, (1980).
- "The Temperament of Lambert Chaumont" J. Acoust. Soc. Am. 67, S85 (1980) (abstract only).
- "Graphic Representations of Complex Tones" J. Acoust. Soc. Am. 67, S98 (1980) (abstract only).
- "Analysis of Zero Sound Attenuation Data for $^3\text{He-A}$ in High Magnetic Fields" (with D. R. Pape) Bull. Am. Phys. Soc. 25, 497 (1980) (abstract only).

WARREN JAMES JOCHEM

Education

M.S., Biomedical Engineering, University of North Carolina, Chapel Hill, N.C., 1982

(Thesis: "An Implantable Hybrid Biotelemetry Transmitter for Detecting Ovulation in Animals")

M.S., Speech and Hearing Sciences, University of North Carolina, Chapel Hill, N.C., 1978

(Thesis: "Phonemic Discrimination Using a Tactile Communication Aid")

B.S., Electrical Engineering, Trenton State College, Trenton, N.J., 1974

(Senior Design Project: An Electronic Intonation Training Aid for the Deaf)

Experience

1982 to date. Research Engineer, Center for Biomedical Engineering, Research Triangle Institute, Research Triangle Park, N.C.

Responsible for designing the analog section of a wearable cardiac monitor, and for coordinating the design of high density circuit boards (Multiwire) for the final prototype. Participated in several clinical studies on non-invasive cardiac monitoring (funded by Duke University and EPA). Participating member of a team studying carboxyhemoglobin buildup in exercising normal subjects exposed to carbon monoxide; activities include data analysis and programming for model validation. Designed a high-efficiency power supply for a portable speech processing aid for the deaf.

1981-1982. Graduate Research Assistant, Biomedical Microelectronics Lab, University of North Carolina, Chapel Hill, N.C.

Assisted in setting up a new thick-film hybrid circuits laboratory and designed an implantable hybrid biotelemetry system.

1978-1981. Research Analyst, Department of Physiology, University of North Carolina, Chapel Hill, N.C.

Designed new instruments for use in electrophysiology laboratories, including an instrument for recording from and marking spinal cord neurons; developed thermal and tactile stimulators; implemented data acquisition systems using LSI-11 computers.

1975-1977. Graduate Research Assistant, Department of Speech and Hearing Sciences, University of North Carolina, Chapel Hill, N.C.

Designed training instruments for speech therapy including a voice pitch/intensity monitor and an instrument to measure nasality.

1975-1977. Student practicum in Audiology, University of North Carolina, Chapel Hill, N.C.

Provided audiological evaluation, hearing aid selection, and patient counselling (300 hours).

1974-1975. Assistant Engineer, Princeton Plasma Physics Laboratory, Princeton, N.J.

Developed low-level RF measurement and fault-detection circuitry for equipment used in nuclear fusion research.

Publications

W. J. Jochem, R. H. Propst, and J. F. Hulka. "An Implantable Hybrid Biotelemetry Transmitter for Detecting Ovulation in Animals," Poster Session presented at the ISHM Symposium, Reno, Nevada, November 1982.

W. J. Jochem, A. R. Light, and D. Smith, "A High-Voltage Electrometer for Recording and Iontophoresis with Fine-Tipped, High-Resistance Microelectrodes," Journal of Neuroscience Methods, 3, 261-269, 1981.

J. Prazma, D. Smith, and W. J. Jochem, "Current to Voltage Converter for Measurement of Oxygen," Journal of Applied Physiology, 44(6), 997-980, 1978.

Professional Associations

IEEE Engineering in Medicine and Biology Society
Biomedical Engineering Society
International Society for Hybrid Microelectronics (ISHM)
Sigma Pi Sigma (National Physics Honor Society)

ROBERT L. BEADLES

Education

B.S., Electrical Engineering, N.C. State University, 1960
M.S., Electrical Engineering, University of Pittsburgh, 1966

Experience

1966 to date. Research Triangle Institute, Research Triangle Park, N.C.

1977 to date. Director, Center for Biomedical Engineering. Responsible for management of interdisciplinary programs in transfer of advanced technology to clinical and research medicine and in biomedical engineering support of research in the health effects of environmental pollution. Performs and directs research in electronic visual prostheses and in automated speech recognition.

1971-1977. Senior Engineer, Center for Technology Applications. Performed and directed research in electronic prostheses and biomedical applications of minicomputers and microprocessors. Principal investigator of a study to investigate feasibility of a wearable speech analyzer as a prosthesis for deaf people. Principal investigator of projects to select and evaluate computer-aided instruction systems for deaf students.

1968-1971. Senior Engineer. Project leader for the research and development of a coherent optical processing system for inspection of integrated circuits on the production line. Research in optical and digital enhancement of biomedical imagery.

1966-1968. Research Engineer. Research in computer languages for automated testing of electronic components. Principal author of state-of-the-art reports on uses of computers in reliability and on interconnection and encapsulation of integrated circuits.

1960-1966. Senior Engineer, Westinghouse Defense and Space Center, Baltimore. Weapons system development and advanced computer research.

Professional Associations

Institute of Electrical and Electronic Engineers
Eta Kappa Nu, Electrical Engineering Honorary
Phi Kappa Phi, Scholastic Honorary

Selected Publications and Presentations

"The Implementation of a Modular Arithmetic Computer with Binary Logic Elements," Proceedings Seventh Military Electronics Convention, Washington, D.C., 1963.

Coauthor. "Integrated Circuit Visual Inspection Using Spatial Filtering," IEEE Transactions on Nuclear Science, vol. NS-17, no. 6, December 1970.

Coauthor. "Automatic Cued Speech," Proceedings Research Conference on Speech-Analyzing Aids for the Deaf, Washington, D.C., May 1977.

Coauthor. "An Interactive Echographic Processor," Twelfth Hawaii International Conference on System Sciences, Honolulu, January 1979.

Inventions

Three patents pending on speech perception prostheses, speech recognition, and low-vision aids.

CURRICULUM VITAE

Mark White, Ph.D.

EDUCATION:

University of Nebraska	1966-1971 B.S.E.E., with High Distinction
University of California, Berkeley	1972-1977 Ph.D., in Electrical Engineering and Computer Science-- Biomedical Engineering

POSITIONS HELD:

Employment Prior to Doctorate

1. Broadcast Engineer. Monitoring and maintenance of AM and FM radio transmitters. KRVN Radio, Lexington, Nebraska, 1966.
2. Television Broadcast Engineer. Maintenance and monitoring of television and audio broadcast facilities. KUON-TV, Lincoln, Nebraska, 1967-1968.
3. Fortran and Cobol Computer Programmer. University of Nebraska Computer Center, Lincoln, Nebraska, 1968.
4. Design Engineer. Developed software for automatically testing computer subassemblies. Collins Radio Corp, Richardson, Texas, 1968.
5. Design Engineer. Designed audio recording equipment. SECO Laboratory, Omaha, 1971.
6. Design Engineer. Designed electronic surveillance equipment. Guard Systems Inc., Oakland, 1971-1972.
7. Research Assistant. Designed, fabricated, and tested a significant portion of a primate oculomotor research laboratory in neurophysiology. Electrical Engineering and Computer Science Dept., UC Berkeley, 1972.
8. Research Assistant. Designed laboratory instrumentation for research in saccadic rhythm. Entomology Dept., UC Berkeley, 1973-1974.

Employment within the University of California, San Francisco

- 1977-1978 Assistant Professor in Residence, Department
 of Otolaryngology
- 1978-1979 Adjunct Assistant Professor, Department of
 Otolaryngology
- 1979-Pres. Assistant Professor in Residence, Department
 of Otolaryngology

HONORS & AWARDS:

- Regents Scholarship
Haskel Scholarship
Eta Kappa Nu Scholarship
National Institutes of Health Traineeship in Biomedical
Graduate Study, 1972-1977
Memberships in honor societies: Sigma Tau, Eta Kappa Nu, and Phi
Eta Sigma
Graduated with High Distinction, University of Nebraska
Co-Chair of Neuroprosthesis Section of the Third International
Congress on Artificial Organs, Paris, July, 1981.

MEMBERSHIP IN SCIENTIFIC SOCIETIES:

- 1976-1980 Member, Bay Area Auditory Research Group
1978-Pres. Member, Institute of Electrical and Electronics Engineers
1980-Pres. Associate Member, Acoustical Society of America
1981-1983 Member, International Society for Artificial Organs
1984-Pres. Association for Research in Otolaryngology (membership
 application pending)

PRESENTATIONS BEFORE SCIENTIFIC SOCIETIES:

- 1976 Research Forum, American Academy of Ophthalmology and
 Otolaryngology, Las Vegas.
 Presentation: Further Progress in the Development of
 Multichannel Cochlear Implants.
- 1977 Annual Meeting, Committee on Hearing, Bioacoustics and
 Biomechanics, National Research Council, Washington, D.C.
 Presentation: Electrophysiological Experiments Relating
 to Electrical Stimulation of the Auditory
 Nerve in Man.
- 1977 Participant in a symposium to consider the feasibility of
 a cochlear implant as a medical benefit of the British
 National Health Service. Sponsored by the British

National Health Service.

- 1978 First International Course on Multichannel Cochlear Implants, Hopital Saint-Antoine, Paris.
Presentation: Some Electrophysiological Data Relavant to (Invited) Excitation Control for a Cochlear Prosthesis.
- 1978 Acoustical Society of America, Waikiki, Hawaii.
Presentation: Multichannel Cochlear Prosthesis; Noninvasive Recording Methods for Estimating the Spatial Distribution of Functional Auditory Nerve Fibers and the Spatial Distribution of Electrically Excited Nerve Fibers.
- 1979 Acoustical Society of America, Salt Lake City.
Presentation: Progress in Electrical Stimulation of the (Invited) Cochlea.
- 1980 West Coast Cochlear Prosthesis Conference, Seattle.
Presentation: Formant Frequency Discrimination in a Subject Implanted with an Intracochlear Stimulating Electrode using Natural and Synthetic Speech.
- 1980 Society for Neuroscience, Cincinnati, Ohio.
Presentation: On the Role of Auditory Nerve Inter-Fiber Discharge Timing in the Perception of Pitch.
- 1980 Acoustical Society of America, Los Angeles.
Presentation: Formant Frequency Discrimination in a Subject Implanted with an Intracochlear Stimulating Electrode.
- 1981 West Coast Cochlear Prosthesis Conference, Pacific Grove, CA.
Presentation: Temporal Properties of Unit Responses in the Anteroventral Cochlear Nucleus due to Intra-Cochlear Electrical Stimulation.
- 1981 Third Congress of the International Society for Artificial Organs, Paris.
Presentation: Evaluation of Two Subjects Implanted with a Multielectrode Intracochlear Implant: Discriminable Features of Speech.
- Co-Chair of Neuroprosthesis Section of the Third International Congress on Artificial Organs.
- 1982 New York Academy of Sciences: International Cochlear Implant Conference.
Presentation: Multi-electrodes: single channel vs. multi-channel results.
(Invited)
- Panel Member: Coding Considerations in the Transmission of Speech Information using a Cochlear Prosthesis.

- 1982 West Coast Cochlear Prosthesis Workshop, Seattle, Washington
Presentation: Methods for Measuring Channel Interaction:
Single Unit, ABR, and Psychophysical Response
Measures.

Moderator: Neurophysiology session.
- 1983 Sixth Midwinter Research Meeting of the Association for
Research in Otolaryngology, St. Petersburg Beach, Florida.
Presentation: Electrical Stimulation of the Auditory Nerve:
Membrane Models Applied to the Interpretation
of Psychophysical and Electrophysiological
Responses.
Presentation: Multichannel Electrical Stimulation of the
(Invited) Auditory Nerve: Channel Interaction and
Processor Design.

Panel Member: Maximizing Auditory Information via
Single vs. Multi-Channel Cochlear
Prostheses.
- 1983 Tenth Anniversary Conference on Cochlear Implants: An
International Symposium, San Francisco, California.
Presentation: Speech Processing Strategies
(Invited)
- 1983 Gordon Research Conference on Cochlear Prostheses,
Tilden, New Hampshire.
Presentation: Neurophysiological and Psychophysical
(Invited) Considerations in the Design of a
Cochlear Prosthesis.
- 1983 First Vienna International Workshop on Functional
Electrostimulation, Vienna, Austria.
Presentation: Compression Systems for Cochlear
Prostheses.
- 1983 Acoustical Society of America, San Diego, California.
Presentation: Electrical Stimulation of the Auditory
Nerve: Membrane Models Applied to the
Interpretation of Electrophysiological
and Psychophysical Responses.
Presentation: Electrical Stimulation of the Auditory
Nerve in Man: Dynamic Range as a Function
of Stimulus Duration.
- 1984 Seventh Midwinter Research Meeting of the Association for
Research in Otolaryngology, St. Petersburg Beach, Florida.
Presentation: Electrical Stimulation of the Auditory
Nerve in Man: Compression Systems.
- 1984 West Coast Cochlear Prosthesis Conference, Seattle
Washington.
Presentation: Channel Interactions as a Function of
Stimulus Amplitude and Waveform.

- 1984 XVII International Congress of Audiology, Santa Barbara, California.
Presentation: The Cochlear Implant: Dynamic Range Functions and Compressors.
Chairman: Cochlear prosthesis session.
- 1984 Sixth Annual Conference of the IEEE Engineering in Medicine and Biology Society.
Presentation: The Multichannel Cochlear Prosthesis: (Invited) Channel Interactions.
- 1984 Acoustical Society of America, Minneapolis, Minnesota.
Presentation: Interactions in a Multichannel Cochlear Prosthesis.

**REGIONAL AND NATIONAL SCIENTIFIC MEETINGS ATTENDED
IN THE LAST THREE YEARS:**

- 1981 West Coast Cochlear Prosthesis Conference, Pacific Grove, CA.
- 1981 Third Congress of the International Society for Artificial Organs, Paris.
- 1982 New York Academy of Sciences: International Cochlear Implant Conference.
- 1982 West Coast Cochlear Prosthesis Workshop, Seattle, Washington.
- 1983 Sixth Midwinter Research Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, Florida.
- 1983 Tenth Anniversary Conference on Cochlear Implants: An International Symposium, San Francisco, California.
- 1983 Gordon Research Conference on Cochlear Prostheses, Tilden, New Hampshire.
- 1983 First Vienna International Workshop on Functional Electrostimulation, Vienna, Austria.
- 1983 Acoustical Society of America, San Diego, California.
- 1984 Seventh Midwinter Research Meeting of the Association for Research in Otolaryngology, St. Petersburg Beach, Florida.
- 1984 West Coast Cochlear Prosthesis Conference, Seattle Washington.
- 1984 XVII International Congress of Audiology, Santa Barbara,

California.

- 1984 Sixth Annual Conference of the IEEE Engineering in
Medicine and Biology Society, Los Angeles, California.
- 1984 Acoustical Society of America, Minneapolis, Minnesota.

SERVICE TO EDITORIAL BOARDS:

- 1979- Reviewer, Journal Biomedical Engineering, Institute of
Pres. Electrical and Electronics Engineers
- 1981 Grant reviewer, Western Regional Veterans Administration.

TEACHING

<u>Speech & Hearing Science Course No. and Title</u>	<u>Grad.- Underg.</u>	<u>Units</u>	<u>Qtrrs per yr.</u>	<u>From</u>	<u>To</u>	<u>Person in Charge</u>
230; Seminar on Research in Speech Reception	Grad.	3	1	1980	Pres	M. White
234; Auditory Psychophysics	Grad.	3	1	1983	Pres	M. White
212; Speech and Hearing Science Seminar	Grad.	1	1	1983	Pres	M. White
250; Dissertation Research	Grad.	1	3	1982	Pres	M. White
299; Dissertation	Grad.	1	3	1982	Pres	M. White
249; Independent Study	Grad.	4	2	1981		M. White
210; Journal Club	Grad.	1	1	1982		R. Snyder
249; Independent Study-	Grad.	2	1	1984		M. White
EECS 109; UC-Berkeley Electrical Eng. & Computer Sci. Dept.; Electronic circuits and Instrumentation Systems	Underg.	4	1	1983		M. White

Summary of Teaching Duties

Total hours of teaching during past academic year	1983-84	208
Total hours of teaching during current year	1984-85	255
Total hours anticipated	1985-86	255

OTHER POSTGRADUATE TEACHING (Extended Programs in Medical Education)

<u>Title of Presentation(s) and Name of Course</u>	<u>Dates Given</u>	<u>Hours of Participation</u>
Electrophysiological Data Relevant to Excitation Control for a Cochlear Prosthesis. First International Course on Multichannel Cochlear Implants, Hopital Saint-Antoine, Paris.	1978	2 hour presentation 3 days participation

The Role of the Engineer in Biomedical Research. Presented to graduating engineers at the University of California, Berkeley.	1978	45 min presentation
The coding of auditory information in the auditory periphery. University of California, San Francisco, ENT Grand Rounds.	1982	1 hr presentation & discussion
Speech Processing Strategies. 10th Anniversary Conference on Cochlear Implants: An International Symposium.	1983	15 min presentation 2 days participation

NAMES OF GRADUATE STUDENTS SUPERVISED

<u>Student</u>	<u>Years Supervised</u>	<u>Type of Supervision</u>
Linda D'Antonio	1980-1982 recvd Ph.D. 1982	(chair, dissertation committee)
Yvonne Sininger	1983-Present	(dissertation committee)
Karen Doyle	1982-1984 recvd Ph.D. 1984	(dissertation committee)
David Morledge	1984-Present	(directing student research)

UNIVERSITY SERVICE

Bioengineering Group

Member, Joint UCSF-UCB Bioengineering Graduate Program, 1983-Present.
 Member, UCSF-UCB Bioengineering Training Grant Committee, 1984.
 Member, UCSF-UCB Bioengineering Nominating Committee, 1984.
 Participated in a 2-day organizational meeting of the newly-formed joint UCSF-UCSB bioengineering group, Asilomar, January, 1984.

Speech and Hearing Science Group

Member, UCSF-UCSB Speech and Hearing Science Group, 1979-Present.
 Graduate Advisor, Speech and Hearing Science Group, 1984-1985.
 Helped to obtain grants for equipment necessary for graduate student training in Speech and Hearing Science: a computer graphics terminal and a software package for signal analysis (1980-1982). This software was integrated with our software to generate a complete laboratory system for the analysis of speech signals by students. Several students have used this system extensively throughout their doctoral research (1981-Present).
 Member, Speech and Hearing Science Curriculum Committee, 1983-Present.

Participated in two joint faculty UCSF-UCSB Speech and Hearing Science organizational meetings at Santa Barbara, 1983 & 1984.

Participated in a program to familiarize UCSB and UCSF students with the educational opportunities available on the UCSF campus, 1984.

Cochlear Protheses Research Group, Dept of Otolaryngology

Executive Committee member, 1977-1979.

Psychophysical Committee member, 1978-Present.

Patient Selection Committee member, 1982-1983.

Engineering Review Committee member, 1984-Present.

Other University Service

Dept of Otolaryngology Faculty Review Committee member, 1979.

Visiting Faculty Member - University of California, Berkeley,

Electrical Engineering and Computer Science Department, 1983.

Member, UCSF Biostatistics Group, 1982-Present.

PUBLIC SERVICE

Participated in an NSF sponsored conference concerning the application and feasibility of cochlear protheses, San Francisco, 1975.

Participated in a 2-day meeting for determining the feasibility of cochlear protheses as a medical benefit of the British National Health Service. The meeting was sponsored by the British National Health Service, 1977.

Participated in the National Research Council's Annual Meeting of the Committee on Hearing, Bioacoustics and Biomechanics, 1977.

Participated in the National Technical Institute for the Deaf workshop on prosthetic devices for the deaf, 1979.

Communicated with representatives of the United States Department of Health and Human Services, 1984. Answered detailed questions concerning the current status of cochlear protheses. Representatives were the Director of OB-GYN, ENT, and Dental Devices, Office of Device Evaluation, Center for Devices and Radiological Health and an associate of the Director.

Signal processing methods which may improve the performance of hearing aids were presented in written form to the engineering staffs of major hearing aid manufacturers: Starkey, Siemens, Maico, Widex, Oticon, and Radioear.

PATENTS AND PATENT APPLICATIONS:

1984 Patent Application: Signal Compression System
Filed with United States Patent Office: July 24, 1984,
United States Patent Office Serial No. 633,943.
University of California Case No. 83-166-1.

PUBLICATIONS:

1. Merzenich, M.M., Schindler, D.N., and White, M.W.:
Symposium on cochlear implants. II. Feasibility of
multichannel scala tympani stimulation. Laryngoscope
84:1887-1893, 1974.
2. Schindler, R.A., Merzenich, M.M., White, M.W., and
Bjorkroth, B.: Multielectrode intracochlear implants:
Nerve survival and stimulation patterns. Arch.
Otololaryngol. 103:691-699, 1977.
3. Merzenich, M.M., White, M., Leake, P.A., Schindler, R.A.,
and Michelson, R.P. Further progress in the development
of multichannel cochlear implants. Trans. Amer. Acad.
Ophth. Otol. 84:181-182, 1977.
4. White, M.W.: Design considerations of a prosthesis for
the profoundly deaf. Doctoral Dissertation, U.C.
Berkeley, 1978.
5. Michelson, R.P., Schubert, E., Walsh, S.W., and White,
M.W.: Protocol for determining the auditory percepts of
electrical stimulation of the cochlea. Laryngoscope
89:748-751, 1979.
6. White, M.W. and Merzenich, M.M.: Aspect electro-
physiologique de l'implant cochléaire a multi-electrodes.
Les Cahiers D'O.R.L. T14 6:567-579, 1979.
7. Merzenich, M.M., White, M.W., Vivion, M.C., Leake-Jones,
P.A., and Walsh, S.W.: Some considerations of multichannel
electrical stimulation of the auditory nerve in the
profoundly deaf: Interfacing electrode arrays with the
auditory nerve array. Acta Otolaryngol. 87:196-203, 1979.
8. Merzenich, M.M., and White, M.W.: Coding considerations
in design of cochlear prostheses. Ann. Otol. Rhinol.
Laryngol. Suppl. 74, 89:84-87, 1980.
9. Merzenich, M.M., Byers, C.L., White, M. and Vivion, M.C.:
Cochlear implant prostheses: Strategies and progress.
Annals Biomedical Engineering 8:361-368, 1980.
10. Vivion, M.C., Merzenich, M.M., Leake-Jones, P.A., White,

M.W., and Silverman, M.: Electrode position and excitation patterns for a model cochlear prosthesis. Ann. Otol., Rhinol. & Laryngol. Suppl. 82, 90:19-20, 1981.

11. Vurek, L.S., White, M.W., Fong, M., Walsh, S.M. Opto-isolated stimulators used for electrically evoked BSER. Ann. Otol., Rhinol., & Laryngol. suppl. 82, 90:21-24, 1981.
12. White, M.W. Formant frequency discrimination in a subject implanted with an intracochlear stimulating electrode. Art. Organs 5S:314-316, 1981.
13. Loeb, G.E., White, M.W., Jenkins, W.M.: Biophysical considerations in electrical stimulation of the auditory system. Annals of the New York Academy of Sciences 405:123-136, 1983.
14. White, M.W., Formant frequency discrimination and recognition in subjects implanted with an intracochlear stimulating electrode. Annals of the New York Academy of Sciences 405:348-359, 1983.
15. White, M.W.: Compression Systems for Cochlear Prostheses. Mechanisms of Hearing ed. W.R. Webster and L.M. Aitkin, Monash Press, Clayton Victoria, Australia; p. 184-189, 1983.
16. Loeb GE, White MW, Merzenich MM. Spatial cross-correlation: A proposed mechanism for acoustic pitch perception. Biol Cybernetics 47:149-163, 1983.
17. White, M.W.: The cochlear implant: the coding of auditory information. Proceedings of the First Vienna International Workshop on Functional Electrostimulation, edited by Dr. H. Thoma, University of Vienna, Vienna, Austria; p. 240-243, 1983.
18. White, M.W., Merzenich, M.M., Gardi, J.N.: Multichannel cochlear implants: channel interactions and processor design. Archives of Otolaryngology 110:493-501, 1984.
19. White, M.W.: The Multichannel Cochlear Prosthesis: Channel Interactions. Proceedings of the Sixth Annual Conference of the IEEE Engineering in Medicine and Biology Society--Frontiers of Engineering and Computing in Health Care, Los Angeles, California, p.396-400, 1984.

IN PRESS:

1. White, M.W.: Psychophysical and Neurophysiological Considerations in the design of a cochlear prosthesis. Italian Journal of Audiology (To be published, December, 1984).

2. White, M.W.: Compression systems for hearing aids and cochlear prostheses. Journal of Rehabilitation Engineering: Research and Development.

SUBMITTED:

1. White, M.W., Merzenich, M.M., Loeb, G.E.: Electrical stimulation of the eighth nerve in cat: temporal properties of unit responses in the large spherical cell region of the AVCN. Ann Otol Rhinol Laryngol.
2. White, M.W.: Cochlear prostheses: Neural and behavioral functions of stimulus intensity. Hearing Research.

BOOKS:

1. Merzenich, M.M. and White, M.W.: Cochlear prosthesis: The interface problem. In: Functional Electrical Stimulation. (Hambrecht, F.T., and Reswich, J.B. eds.), Marcel Dekker, Inc., Vol 3, pp. 321-340, 1977.
2. White, M.W., Merzenich, M.M. and Vivion, M.C.: A non-invasive recording technique for evaluating electrode and nerve function in cochleas implanted with a multichannel electrode array. In: Advances in Prosthetic Devices for the Deaf: A Technical Workshop. (McPherson, D.L. ed.), The National Technical Institute for the Deaf, Rochester Institute of Technology, Rochester, N.Y., pp. 330-334, 1980.
3. Merzenich, M.M., Vivion, M.C., Leake-Jones, P.A. and White M.W.: Progress in development of implantable multielectrode scala tympani arrays for a cochlear implant prosthesis. In: Advances in Prosthetic Devices for the Deaf: A Technical Workshop. (McPherson, D.L. ed.), The National Technical Institute for the Deaf, Rochester Institute of Technology, Rochester, N.Y., pp. 262-270, 1980.

IN PRESS

1. White, M.W.: Speech and stimulus processing strategies for cochlear prostheses. In: Cochlear Implants: Current Status and Future. (R.A. Schindler and M.M. Merzenich, eds.) Raven Press, New York, (To be published, Dec. 11, 1984).

TECHNICAL REPORTS

1. Merzenich, M.M., Jones, P., White, M., Vivion, M., Silverman, M.: Development of multichannel electrodes for an auditory prosthesis - first quarterly progress report (Sept.-Nov. 1977). Report available through NIH, contract # N01-NS-7-2367.
2. Merzenich, M.M., Vivion, M., Jones, P., White, M., Mcmillan, B., Silverman, M.: Development of multichannel electrodes for an auditory prosthesis - second quarterly progress report (Dec. 1977-Feb. 1978). Report available through NIH, contract # N01-NS-7-2367.
3. Merzenich, M.M., Vivion, M., Jones, P., White, M., Silverman, M.: Development of multichannel electrodes for an auditory prosthesis - fourth quarterly progress report (June-Aug. 1978). Report available through NIH, contract # N01-NS-7-2367.
4. Merzenich, M.M., Byers, C., Jones, P., Rebscher S., Walsh, S.M., White, M.: Development of multichannel electrodes for an auditory prosthesis - yearly progress report for 1979-1980. Report available through NIH, contract # N01-NS-7-2367.
5. Merzenich, M.M., White, M., Shannon, R.V., Gray, R.F., Byers, C., Rebscher S., Casey, D.E.: Development of multichannel electrodes for an auditory prosthesis - quarterly progress report (Sept.-Nov. 1981). Report available through NIH, contract # N01-NS-7-2367.
6. Rebscher, S.J., Wilkinson, D.R., Zimmerman, P.A., Byers, C.L., Merzenich, M.M., White, M.: Development of multichannel electrodes for an auditory prosthesis - 3rd quarterly progress report (April-May 1984). Report available through NIH, contract # N01-NS-3-2353.

ABSTRACTS:

1. Merzenich, M.M., White, M.W., Schindler, R.A., Schindler, D.N. and Michelson, R.P.: Progress in the development of a multi-channel acoustic nerve stimulation prosthesis for the totally deaf. J. Acoust. Soc. Am. 57:S72, 1975.
2. Vivion, M.C., Merzenich, M.M., Leake-Jones, P.A., and White, M.W. Defining optimum electrode arrays for multichannel cochlear prostheses: electrophysiological studies. J. Acoust. Soc. Am. 64:S65, 1978.

3. White, M.W., Merzenich, M.M. and Vivion, M.C.: Multichannel cochlear prosthesis: Non-invasive recording methods for estimating the spatial distribution of electrically excited nerve fibers. J. Acoust. Soc. Am. 64:S64, 1978.
4. White, M.W.: Progress in electrical stimulation of the cochlea. J. Acoust. Soc. Am. 66:S17, 1979.
5. White, M.W. Formant frequency discrimination in a subject implanted with an intracochlear stimulating electrode. J. Acoust. Soc. Am. 68:S44, 1980.
6. White, M.W., Loeb, G.E. and Merzenich, M.M.: On the role of auditory nerve inter-fiber discharge timing in the perception of pitch. Soc. Neuroscience Abstrs. 6:552, 1980.
7. Loeb, G.E., Merzenich, M.M. and White, M.W.: Brainstem processing of auditory information to generate a central representation of pitch. Soc. Neuroscience Abstrs. 6:552, 1980.
8. Merzenich, M.M., Loeb, G.E. and White M.W. Extraction of spectral information in auditory brainstem nuclei; hypothesis and experimental observations. J. Acoust. Soc. Am. 68:S1:19, 1980.
9. White, M.W., Merzenich, M.M. and Loeb, G.E. Electrical stimulation of the eighth nerve in cat: Temporal properties of unit responses in the large spherical cell region of the anteroventral cochlear nucleus. J. Acoust. Soc. Am. 69:S1: 97, 1981.
10. Loeb, G.E., White, M.W. and Merzenich, M.M. Mechanisms of auditory information processing for pitch perception. Soc. Neuroscience Abstrs. 7:56, 1981.
11. White, M.W. Evaluation of two subjects implanted with a multielectrode intracochlear implant: a review. Art. Organs 5A:78, 1981.
12. White, M.W., Merzenich, M.M.: Electrical stimulation of the auditory nerve: membrane models applied to the interpretation of electrophysiological and psychophysical responses. J. Acoust. Soc. Am. 72:S6, 1982.
13. Gardi JN, White MW, Merzenich MM. Human brain stem and middle latency responses to electrical stimulation: Preliminary observations. Abstracts of the Sixth Midwinter Research Meeting for Research in Otolaryngology; Jan. 23-27, 1983, St. Petersburg Beach, Florida; p. 24.
14. White, M.W., Merzenich, M.M.: Multichannel electrical

- stimulation of the auditory nerve: channel interaction and processor design. Abstracts of the Sixth Midwinter Research Meeting for Research in Otolaryngology; Jan. 23-27, 1983, St. Petersburg Beach, Florida; p. 128.
15. White, M.W., Merzenich, M.: Electrical stimulation of the auditory nerve: membrane models applied to the interpretation of psychophysical and electrophysiological responses. Abstracts of the Sixth Midwinter Research Meeting for Research in Otolaryngology; Jan. 23-27, 1983, St. Petersburg Beach, Florida; p. 28.
 16. White, M.W.: Speech processing strategies. Abstracts of Tenth Anniversary Conference on Cochlear Implants: An International Symposium; June 22-24, 1983, University of California, San Francisco, Calif; p. 30.
 17. Loeb, G.E., White, M.W., Merzenich, M.M.: Synchronicity Pitch - A neuronal model for extraction of phase-locked activity. In: Mechanisms of Hearing edited by William R. Webster and Lindsay M. Aitkin. Monash Press, Melbourne, Australia.
 18. Jewett, D.L., Shay, D., White, M., Gardi, J.N.: Planar analysis of the 3-channel lissajous trajectory of the ABR: Theoretical basis of planarity. Abstracts of the 8th Biennial International Evoked Response Audiometry Study Group; July 4-8, 1983, Ottawa, Canada pp. 45-46.
 19. White M.W.: Electrical stimulation of the auditory nerve: Membrane models applied to the interpretation of electrophysiological and psychophysical responses. J. Acoust. Soc. Am. 74:S8, 1983.
 20. White, M.W.: Electrical stimulation of the auditory nerve in man: Dynamic range as a function of stimulus duration. J. Acoust. Soc. Am. 74:S110, 1983.
 21. White, M.W.: The multichannel cochlear prosthesis: channel interactions. IEEE Trans. on Biomed. Engin. 31:571, 1984.
 22. White, M.W.: Cochlear implants: Psychophysics related to processor design. Asha 26(no. 10):70, 1984.
 23. White, M.W.: Interactions in a multichannel cochlear prosthesis. J. Acoust. Soc. Am. 76:S48, 1984.
 24. White, M.W.: Electrical stimulation of the auditory nerve in man: Compression systems. Abstracts of the Seventh Midwinter Research Meeting for Research in Otolaryngology February 5-9, 1984, St. Petersburg Beach, Florida; p. 100.
 25. White, M.W.: The cochlear implant: Dynamic range functions and compressors. Abstracts of the 17th

International Congress of Audiology August 26-30, 1984,
University of California, Santa Barbara, CA; p. 103.

26. White, M.W.: Channel interactions as a function of stimulus amplitude and waveform. Abstracts of the 1984 West Coast Cochlear Prosthesis Workshop July 29-31, 1984, Battelle Institute, University of Washington, Seattle, Washington; p. 11.

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BIOGRAPHICAL DATA

Born: May 19, 1927 (Fargo, North Dakota)
Married: Judith Johnson (August, 1965)
Children: Jill (August 1969) and Bradley (May, 1972)

EDUCATION

<u>Degree</u>	<u>Institution</u>	<u>Date</u>	<u>Major</u>
B.A.	Concordia College	1959	Psychology
M.S.	Pennsylvania State University	1961	Audiology
Ph.D.	University of Illinois	1966	Audiology

PROFESSIONAL EXPERIENCE

June 1960 - May 1962 Audiologist, Michigan Association
for Better Hearing - East Lansing,
Michigan

June 1962 - August 1963 Audiologist and Instructor, Depart-
ment of Speech, Central Michigan
University, Mt. Pleasant, Michigan

September 1966 - August 1971 Assistant Professor, Department of
Speech, University of Michigan, Ann
Arbor, Michigan

September 1971 - August 1977 Associate Professor, Department of
Speech and Hearing Sciences, Univer-
sity of Washington, Seattle,
Washington

Chairman, Communication Disorders
Research Program
Child Development and Mental
Retardation Center

Head, Audiology Division
Department of Speech and Hearing
Sciences

September 1977 - Present

Associate Professor of Audiology
Department of Surgery
Center for Speech and Hearing
Disorders
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Coordinator of Audiology Section

OTHER CURRENT PROFESSIONAL ACTIVITIES

Member, Board of Directors of North Carolina Speech-Hearing-Language Association (Counselor for Clinical Services).

Section Editor for Ear and Hearing (Electrophysiologic Techniques in Audiology and Otology)

Education and Training Board Site Visitor for the American Speech and Hearing Association.

Member, Panel of Judges for the Beltone Distinguished Teaching Award in Audiology.

Consultant to Division on Exceptional Children, North Carolina Department of Public Instruction.

Fellow, American Speech-Hearing-Language Association

Member, North Carolina Speech-Hearing-Language Association

Member, Editorial Board, American Auditory Society

Member, International Evoked Response Audiometry Study Group

Member, Sigma Xi

LICENSES

North Carolina Audiology License 1977-present

Certification in Clinical Competency in Audiology by American Speech-Hearing-Language Association 1972-present.

PUBLICATIONS

- Webb, C., S. Kinde, B. Weber and R. Beedle, Incidence of Hearing Loss in Institutionalized Mental Retardates. American Journal of Mental Deficiency, 70: 563-568, 1966.
- Beedle, R., C. Webb, S. Kinde and B. Weber, A Behavioral Observation Technique of Hearing Screening of Institutionalized Mentally Retarded Children. American Journal of Mental Deficiency, 70: 675-682, 1966.
- Weber, B.A., and W. Milburn, The Effects of Rhythmically-Moving Auditory Stimulus on Eye Movements in Normal Young Adults. Journal of Auditory Research, 7: 259-266, 1967.
- Weber, B.A., The Use of Involuntary Hearing Testing Procedures with Children. Journal of Communication Pathology, 1: 58-61, 1968.
- Weber, B.A., Validation of Observer Judgments in Behavioral Observation Audiometry. Journal of Speech and Hearing Disorders, 34: 350-355, 1969.
- McFarland, W., and B.A. Weber, An Investigation of Ocular Movements to Various Forms of Sound Field Auditory Stimulation. Journal of Auditory Research, 9: 236-239, 1969.
- Weber, B.A., Habituation and Dishabituation of the Averaged Auditory Evoked Response. Journal of Speech and Hearing Research, 13: 387-394, 1970.
- Weber, B.A., A Comparison of Two Approaches to Behavioral Observation Audiometry. Journal of Speech and Hearing Research, 13:823-825, 1970.
- Sulzbacher, S.I., B.A. Weber and A.R. Biles, Direct Measurement of Drug Effects on the Behavior of Children: Electroencephalic Response and Schedule-Dependent Operant Behavior. Clinical Research, 19: 231, 1971.
- Weber, B.A., The Use of the AEA with Infants. Journal of Speech and Hearing Disorders, 37: 142-143, 1972.
- Weber, B.A., Short-Term Habituation of the Averaged Electroencephalic Response in Infants. Journal of Speech and Hearing Research, 15: 757-762, 1972.
- McSpaden, J., and B.A. Weber, Averaged Electro-Ocular Responses to Acoustic Stimuli. Journal of Auditory Research, 12, 76-81, 1972.

PUBLICATIONS (cont'd)

- Weber, B.A., The Current Status of Averaged Electroencephalic Audiometry with Children. Clinical Pediatrics, 12:629-630, 1973.
- Weber, B.A., and M. Dybka, The Use of the Averaged Electroencephalic Response in the Study of Auditory Discrimination. Journal of Auditory Research, 13: 45-49, 1973.
- Thompson, G., and B.A. Weber, Responses of Infants and Young Children to Behavior Observation Audiometry. Journal of Speech and Hearing Disorders, 39: 140-147, 1974.
- Seitz, M.R., and B.A. Weber, Effects of Response Requirements on the location of Clicks Superimposed on Sentences. Memory and Cognition, 1: 43-46, 1974.
- Ilecki, H.J., and B.A. Weber, The Averaged Electroencephalic Response in the Discrimination of Complex Auditory Stimuli. Journal of Auditory Research, 14: 279-282, 1974.
- D.E. Anderson and B.A. Weber, Cortical Responses of Infants to Selected Acoustic Patterns. Journal of Auditory Research, 14: 195-199, 1974.
- Weber, B.A., and S.I. Sulzbacher, Use of CNS Stimulant Medication in Averaged Electroencephalic Audiometry with Children with Minimal Brain Dysfunction. Journal of Learning Disabilities, 8: 300-304, 1975.
- Helmick, J.W., and B.A. Weber, Effects of Stimulus Form on Strategies Used in Concept Identification. Acta Symbolica, 6: 19-26, 1976.
- T.N. Decker and B.A. Weber, The Effects of Subject Listening State on Potentials Associated with Missing Auditory Stimuli, Journal of Auditory Research. 16: 177-181, 1976.
- Weber, B.A., and R.C. Folsom, Brainstem Wave V Latencies to Tone Pip Stimuli. Journal of the American Audiology Society, 2: 182-184, 1977.
- Weber, B.A., and G.S. Omenn, Auditory and Visual Evoked Responses in Children with Familial Reading Disabilities. Journal of Learning Disabilities, 10: 153-158, 1977.
- Weber, B.A., and S.M. Fujikawa, Brainstem Evoked Response (BER) Audiometry at Various Presentation Rates. Journal of Learning Disabilities, 10: 153-158, 1977.

PUBLICATIONS (Cont'd)

- Bernard, K., W. Wenner, B. Weber, C. Gray and A. Peterson, Infant Refocus in Research to Practice in Mental Retardation: Biomedical Aspects, Vol. III, P. Mittler (Ed.), I.A.S.S.M.D., 191-198, 1977.
- Fujikawa, S.M., and B.A. Weber, Effects of Increased Stimulus Rate on Brainstem Electric Response (BER) Audiometry as a Function of Age. Journal of the American Audiology Society, 3: 147-150, 1977.
- Weber, B.A., Brainstem Electric Response Audiometry: Some Practical Considerations. Journal of the North Carolina Speech, Hearing and Language Association, 8: 19-24, 1978.
- Omenn, G.S., and B.A. Weber, Dyslexia: Search for Phenotypic and Genetic Heterogeneity. American Journal of Medical Genetics, 1: 333-342, 1978.
- Weber, B.A., Auditory Brain-Stem Response Audiometry in Children. Clinical Pediatrics, 18: 746-749, 1979.
- Weber, B.A., Spalding, J.L.P., and Fletcher, G.L., Auditory Brainstem Response Audiometry: Cautions and Practical Considerations. Hearing Aid Journal, 33:6, 40-42, 1980.
- Seitz, M.R., Weber, B.A., Jacobson, J.T., and Morehouse, R., The Use of Averaged Electroencephalic Response Techniques in the Study of Auditory Processing Related to Speech and Language. Brain and Language, 11:261-284, 1980.
- Weber, B.A., and Fletcher, G.L., A Computerized Scoring Procedure for Auditory Brainstem Response Audiometry. Ear and Hearing, 1: 233-236, 1980.
- Weber, B.A., Seitz, M.R., and McCutcheon, M.J., Quantifying Click Stimuli in Auditory Brainstem Response Audiometry. Ear and Hearing, 2: 15-19, 1981.
- Fuller, P.W., Weber, B.A., and Fujikawa, S.M., The Averaged Evoked Potential in the Study of Infant Auditory Discrimination. Child Development, 52: 749-751, 1981.
- Weiner, R.D., Erwin, C.W., and Weber, B.A., Acute Effects of Electroconvulsive Therapy on Brain Stem Auditory-Evoked Potentials. Electroencephalography and Clinical Neurophysiology, 52: 202-204, 1981.
- Weber, B.A., Comparison of Auditory Brain Stem Response Latency Norms for Premature Infants. Ear and Hearing, 3: 257-262, 1982.

PUBLICATIONS (Cont'd)

- Eccard, K.E., and Weber, B.A., Influence of Electrode Impedance on ABR Recordings in the Intensive Care Nursery. Ear and Hearing, 4: 104-105, 1982.
- Folsom, R.C., Weber, B.A., and Thompson, G., Auditory Brainstem Responses in Children with Early Recurrent Middle Ear Disease. Annals Otolaryngology, Rhinology and Laryngology, 92: 249-253, 1983.
- Weber, B.A., Pitfalls in Auditory Brain Stem Response Audiometry, Ear and Hearing, 4: 179-184, 1983.
- Weber, B.A., Masking and Bone Conduction Testing in Brainstem Response Audiometry. Seminars in Hearing, 4: 343-352, 1983.
- Hooks, R.G. and Weber, B.A., Auditory Brainstem Responses of Premature Infants to Bone-Conducted Stimuli: a Feasibility Study. Ear and Hearing, 5: 42-46, 1984.
- Weber, B.A., Interpretation: Problems and Pitfalls in The Auditory Brainstem Response (J.T. Jacobson, Ed), San Diego, College-Hill Press, 1984.

CONVENTION PAPERS

- "Audiogenic Eye Movements in Normal Young Adults." (With M. Milburn)
American Speech and Hearing Association Convention, Chicago, IL,
1967.
- "Observer Objectivity in Distraction Audiometry with Children."
American Speech and Hearing Association, Denver, CO, 1968.
- "Effects in Two CNS Stimulants on the Auditory Evoked Response in
Children." (With S.I. Sulzbacher and R.O. Code)
American Speech and Hearing Association Convention, New York,
N.Y., 1970.
- "Direct Measurement of Drug Effects on the Behavior of Children;
Electroencephalic Responses and Schedule-Dependent Operant
Behavior. (With S.I. Sulzbacher) Western Society for
Pediatric Research. Annual Meeting, Carmel, CA., 1971.
- "Left Hemisphere Processing of Clicks Related to Linguistic
Function." (With M.R. Seitz) American Speech and Hearing
Association Convention, San Francisco, CA., 1972.
- "The Use of the Averaged Electroencephalic response (AER) in the
Study of Auditory Discrimination." (With M.E. Dypka) American
Speech and Hearing Association Convention, Detroit, MI., 1973.
- "The Auditory Evoked Response as an Index of Discrimination of
Complex Auditory Stimuli." (With H.J. Ilecki) American Speech
and Hearing Association Convention, Las Vegas, NV., 1974.
- "Cortical Responses of Infants to Selected Acoustic Patterns."
(With D.E. Anderson) American Speech and Hearing Association
Convention, Las Vegas, NV., 1974.
- "Brainstem Evoked Response (BER) Audiometry: Tool or Toy?" Invited
two hour seminar, Washington Speech and Hearing Association
Convention, Spokane, WA., 1976.
- "Brainstem ERA: Current Procedures." Invited three hour short
course, Canadian Speech and Hearing Association Convention,
Victoria, B.C., Canada, 1977.
- "Measuring Auditory Attending Behavior in Children." (With L.E.
Augustine) American Speech and Hearing Association Convention,
Chicago, IL., 1977.
- "Brainstem Evoked Response Audiometry." Duke-McPherson Hospital
Otolaryngology Symposium, Durham, NC, 1978.
- "Neurophysiology of Auditory Evoked Potentials." Evoked Potential
Symposium for Neurologists, Durham, NC., 1979.

CONVENTION PAPERS (Cont'd)

- "Behavioral and Electrophysiologic Measures of Central Auditory Function." (With C. Michaud) Invited four hour short course, North Carolina Speech, Hearing and Language Association Convention, Wilmington, NC., 1979.
- "Using Correlation to Analyze Brainstem Evoked Potentials." International Electric Response Symposium, Santa Barbara, CA., 1979.
- "Assessment of Hearing in the Infant." Angus M. McBryde Perinatal Symposium, Durham, NC., 1979.
- A Computerized Scoring Procedure for Brainstem Evoked Response Audiometry." (With G. Fletcher) American Speech and Hearing Association Convention, Atlanta, GA., 1979.
- "Sex Differences in Auditory Brainstem Responses (ABRs)." (With M. Seitz and J. Jacobson) International Neuropsychological Society Convention, San Francisco, CA., 1980.
- "Acute Effects of Electroconvulsive Therapy upon Brainstem Auditory Evoked Responses." (With R.C. Weiner and C.W. Erwin) American EEG Society Convention, Boston MA., 1980.
- "Auditory Brainstem Responses in Children with Early Middle Ear Disease." (With R.C.Folsom and G. Thompson). American Speech and Hearing Association Convention, Detroit, Michigan, 1980.
- "Auditory Evoked Potentials: Contribution of Peripheral Mechanisms." Evoked Potential Symposium, Durham, NC., 1981.
- "A Comparison of ABR Latency Norms for Newborns." (With M.D.Menard, O. Murnane and K. Eccard). International Electric Response Audiometry Symposium, Bergamo, Italy, 1981.
- "Clinical Problems in Auditory Brainstem Response Audiometry." North Carolina Speech, Hearing and Language Association, Asheville, NC, 1981.
- "Effects of Recurrent Otitis Media on the Development of Central Auditory Pathways." 16th Annual Duke McPherson Symposium, Durham, NC, 1982.
- "Middle Ear Pathology in Infancy -- the Use of Auditory Brainstem Response (ABR)." Conference on Team Treatment of Cleft Lip and Palate, Durham, NC, 1982.
- "Auditory Brainstem Responses of Premature Infants to Bone Conduction Stimuli." (With R.G. Hooks, G.W. Hume and G.D. Givens). American Speech, Hearing, Language Association Convention, Cincinnati, Ohio, 1983.

CONVENTION PAPERS (Cont'd)

"Auditory Evoked Potentials: Contribution of Peripheral Mechanisms."
Evoked Potential Symposium, Durham, NC, 1983.

"Implications of Recurrent Middle Ear Problems." Conference of Cleft
Palate: Evaluation and Management, Durham, NC, 1983.

WORKSHOPS AND SHORT COURSES CONDUCTED

- Clinical Applications of Auditory Brainstem Response, Saint Josephs Hospital. Savannah, GA., 1981.
- Auditory Evoked Potentials, Landstuhl Army Hospital, Landstuhl, Germany, 1981.
- Auditory Evoked Brain Stem Potentials: Clinical Applications, Wayne State University School of Medicine, Detroit, MI, 1981.
- Auditory Brainstem Response Testing: Clinical Applications to Pediatric Populations, Annual Meeting of Speech-Language-Hearing Association of Virginia, Roanoke, Va, 1982.
- Use of Auditory Brainstem Response Audiometry in Site of Lesion Testing, University of Virginia Medical Center, Charlottesville, Va., 1982.
- The Use of Auditory Brainstem Response Audiometry with Infants and Young Children, Annual Meeting of the West Virginia Speech and Hearing Association, Charleston, WVA, 1983.
- Uses and Abuses of Evoked Auditory Potentials, Annual Meeting of the California Speech-Language-Hearing Association, Palo Alto, CA, 1984.
- BSER Computer Applications and Update, Convention of the Audiological Resource Association, Gatlinburg, TN, 1984.
- Curent Status of ABR Audiometry with Children and Adults, Fifth Annual University of Tennessee Speech, Language, Hearing Summer Convergence, Knoxville, TN, 1984.
- Brainstem, Evoked Response Audiometry for Children and Adults, Annual Meeting of the Kansas Speech and Hearing Association, Hutchinson, Kansas, 1984.

CURRICULUM VITAE

Margaret Walker Skinner
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Born: February 13, 1935; Washington, D.C.

Education

Wellesley College, Wellesley, Massachusetts	A.B. 1956	Chemistry
Case-Western Reserve University, Cleveland, Ohio	M.A. 1960	Audiology
Washington University, St. Louis, Missouri	Ph.D. 1976	Audiology

Honors

Wellesley Scholar, 1955-56
Clinical Traineeship, 1959-60
U.S. Public Health Service Predoctoral Trainee in Medical Audiology,
1971-76

Certification and Licensure

Certificate of Clinical Competence in Audiology, American Speech and
Hearing Association, 1966
License to practice as Clinical Audiologist, State Board for the Healing
Arts of Missouri
License to practice as a Hearing Aid Dealer and Fitter, State of
Missouri, Division of Professional Registration

Research and/or Professional Experience

Director of Audiological Services, Department of Otolaryngology,
Washington University Medical School, 1983-present
Research Associate, Central Institute for the Deaf, St. Louis, Missouri,
1977 to 1983.
U.S. Public Health Service Predoctoral Trainee in Medical Audiology,
Central Institute for the Deaf, St. Louis, Missouri, 1971-1977.
Audiologist, Central Institute for the Deaf, St. Louis, Missouri, 1969-
1971.
Consultant in Audiology, Department of Otolaryngology, All-India
Institute of Medical Sciences, New Delhi, India, 1966-1967.
Audiologist, Department of Otorhinolaryngology, University of Michigan
Medical Center, Ann Arbor, Michigan, 1962-1966.
Teacher and Speech Therapist, Children's Psychiatric Hospital School,
University of Michigan Medical Center, Ann Arbor, Michigan, 1961-
1962.
Audiologist, Cleveland Hearing and Speech Center, Cleveland, Ohio,
1960-1961.
Administrative Aide, U.S. National Bureau of Standards, Washington,
D.C., 1956-1957.

University Affiliations

Assistant Professor of Audiology, Department of Otolaryngology,
Washington University Medical School, 1983-present.
Assistant Professor, Speech and Hearing Department, Washington
University, 1979-1985.
Courses: 561 Hearing Evaluation and Diagnosis

562 Hearing Evaluation and Diagnosis

566 Advanced Hearing Evaluation and Diagnosis

Student advising (one or two students each year).

Sponsor for independent study (one or two students each year).

Lectures on audiological evaluation and hearing aids for the otolaryngology residents.

Lecturer, Speech and Hearing Department, Washington University, 1977-1979. (Same responsibilities as noted above.)

Audiologist, Department of Otorhinolaryngology, University of Michigan Medical Center, Ann Arbor, Michigan, 1962-1966.

Lectured to junior and senior medical students about audiological evaluation.

Presented audiologic results for staff conferences.

Consultant for National Institutes of Health

Purdue University Research Associates contract N01-NS-0-2329, "Determination of Effects of Hearing Aid Amplification on Children," 1979-1983.

Societies and Committees

Acoustical Society of America, member from 1974 to present.

American Speech-Language-Hearing Association, member from 1960 to present;

Member of the Committee on Audiologic Standards 1982-1986.

Member of Internal Review Board, Washington University School of Medicine.

Editorial Review

Review articles for the Journal of the Acoustical Society of America, the Journal of Speech and Hearing Research, the Volta Review, the Laryngoscope, the Journal of Speech and Hearing Disorders, the Annals of Otology, Rhinology, and Laryngology.

Publications

Skinner, M.W. (1976). Speech intelligibility in noise-induced hearing loss: Effects of high-frequency compensation. Ph.D. Dissertation, Washington University, St. Louis, Missouri.

Skinner, M.W. (1978). Hearing of speech during language acquisition. Otolaryngologic Clinics of North America, 11:631-650.

Skinner, M.W. (1979). Audibility and intelligibility of speech for listeners with sensorineural hearing loss. In: Rehabilitation Strategies for Sensorineural Hearing Loss (P. Yanick, ed.) New York: Grune and Stratton, 159-184.

Miller, J.D., Niemoeller, A.F., Pascoe, D., Skinner, M.W. (1980). Integration of the electroacoustic description of hearing aids with the audiologic description of clients. In: Acoustical Factors Affecting Hearing Aid Performance (G.A. Studebaker and T. Hochberg, eds.) Baltimore: University Park Press, 355-377.

Skinner, M.W. (1980). Speech intelligibility in noise-induced hearing loss: Effects of high-frequency compensation. J. Acoust. Soc. Am. 67:306-317.

- Skinner, M.W., Karstaedt, M.M., and Miller, J.D. (1982). Amplification bandwidth and speech intelligibility for two listeners with sensorineural hearing loss. Audiology 21:251-268.
- Skinner, M.W., Pascoe, D.P., Miller, J.D., and Popelka, G.R. (1982). Measurements to determine the optimal placement of speech energy within the listener's auditory area: A basis for selecting amplification characteristics. In: The Vanderbilt Hearing Aid Report (G.A. Studebaker and F.H. Bess, eds.) Upper Darby, Pa.: Monographs in Contemporary Audiology.
- Skinner, M.W. (1982). Effects of peripheral hearing loss on hearing of speech during language acquisition. Seminars in Speech, Language and Hearing 3: 281-294.
- Skinner, M.W., and Miller, J.D. (1983). Amplification bandwidth and intelligibility of speech in quiet and noise for listeners with sensorineural hearing loss, Audiology 22: 253-279.
- Sieger, A., White, N.H., Skinner, M.W., and Spector, G.J. (1983). Auditory function in children with diabetes mellitus. Ann. Oto. Rhin. Laryng. 92: 237-241.
- Skinner, M.W. (1985). Recent advances in hearing aid selection and adjustment, Ann. Oto. Rhin. Laryng. (in press).
- Skinner, M.W., Miller, J.D., DeFilippo, C.F., Dawson, J., and Popelka, G.R. (1985). Word identification by listeners with sensorineural hearing loss using four amplification systems, in Sensorineural Hearing Loss: Mechanisms, Diagnosis and Treatment, to be published by University Park Press.

Invited Talks

- Skinner, M.W., and Pascoe, D.P. (1979). Audibility and intelligibility of speech for hearing impaired listeners, presented at the Second Symposium on the Application of Signal Processing Concepts to Hearing Aids, University Park, Pennsylvania.
- Skinner, M.W. (1979). A procedure for hearing aid selection, presented at the Seminar on Hearing Loss and Sensory Aids for the Elderly, sponsored by the National Institute on Aging and the National Institute of Neurological and Communicative Disorders and Stroke, Bethesda, Maryland.
- Skinner, M.W. (1980). Hearing of speech during language acquisition: effect of conductive and sensorineural hearing loss, presented at the Annual Convention of the Missouri Speech-Language-Hearing Association, Jefferson City, Missouri.

- Skinner, M.W. (1980). Signal degradation and the impaired listener, presented at the Conference on Acoustic Signal Processing sponsored by the Communicative Disorders Program of the National Institute of Neurological and Communicative Disorders and Stroke, Bethesda, Maryland.
- Skinner, M.W. (1981). Hearing aid evaluation and application, presented at Grand Rounds, Department of Otolaryngology, Washington University Medical School, St. Louis, Missouri.
- Skinner, M.W. (1982). Adjusting hearing aids to match amplified speech and residual hearing, presented at the Annual Conference in the Mt. Sinai Series on Communication Disorders, Aural Rehabilitation: Clinical Issues, New York, New York.
- Skinner, M.W. (1982). Effect of amplification bandwidth on speech-intelligibility for hearing-impaired listeners, presented to the Department of Speech and Hearing, Memphis State University, Memphis, Tennessee.
- Skinner, M.W. (1982). Review of research related to hearing aid selection procedures, presented at the Audiology/Hearing Aid Workshop sponsored by the Missouri Speech-Language-Hearing Association, St. Louis, Missouri.
- Skinner, M.W. (1983). Review of research on optimizing speech intelligibility with amplification characteristics. Hearing Aid Workshop, Central Institute for the Deaf, St. Louis, Missouri.
- Skinner, M.W. (1983). Strategies and decision making in hearing aid selection. Section C: The Pascoe Procedure, presented at the American Speech-Language-Hearing Association in Cincinnati, Ohio.
- Skinner, M.W. (1984). Hearing aid selection and adjustment, presented Otolaryngology Update: 1984, Washington University Medical School, St. Louis, Missouri.
- Skinner, M.W. (1984). Word identification by listeners with sensorineural hearing loss using four amplification systems, presented at the Scott Reger Memorial Conference, University of Iowa, Iowa City, Iowa.
- Skinner, M.W. (1984). Recent advances in hearing aid selection and adjustment, presented to the American Otological Society, Palm Beach, Florida.
- Skinner, M.W. (1984). Description of the 3M/Vienna Extracochlear Implant and its Clinical Trial, presented at Grand Rounds, Department of Otolaryngology, Washington University Medical School, St. Louis, Missouri.

Skinner, M.W. (1984). Comparison of five prescriptive procedures for selecting the frequency-gain characteristics of hearing aids, presented to the Division of Speech Pathology and Audiology, Albert Einstein School of Medicine, New York, New York.

RESUME

NAME:

A. Maynard Engebretson

BIRTHDATE:

June 17, 1934

EDUCATION:

University of Minnesota, Minneapolis, Minnesota, B.E.E., 1958, Electrical Engineering.

Washington University, St. Louis, Missouri, M.S.E.E., 1963, Electrical Engineering.

Washington University, St. Louis, Missouri, D.Sc., 1970, Electrical and Biomedical Engineering.

RESEARCH AND/OR PROFESSIONAL EXPERIENCE:

1958-1960 Engineer, McDonnell Aircraft Corporation, St. Louis, MO.
1960-1964 Research Engineer, Central Institute for the Deaf, St. Louis, MO.
1964-1969 Research Assistant, Biomedical Computer Laboratory, Washington University Medical School, St. Louis, MO.
1969-1972 Consultant, General Credit Services, St. Louis, MO.
1972-present Research Associate, Biomedical Computer Laboratory, Washington University Medical School, St. Louis, MO.
1972-1981 Assistant Professor of Electrical Engineering, Department of Speech and Hearing, Washington University, St. Louis, MO.
1972-1983 Research Associate and Head of Digital Methods Laboratory, Central Institute for the Deaf, St. Louis, MO.
1980-1982 Assistant Affiliate Professor, Department of Computer Science, Washington University, St. Louis, MO.
1981-present Associate Professor of Electrical Engineering, Department of Speech and Hearing, Washington University, St. Louis, MO.
1982-present Associate Affiliate Professor, Department of Computer Science, Washington University, St. Louis, MO.
1983-present Senior Research Scientist and Head of Digital Methods Laboratory, Central Institute for the Deaf, St. Louis, MO.
1983-present Assistant Director of Research in Engineering, Central Institute for the Deaf, St. Louis, MO.

PUBLICATIONS:

Engebretson, A.M. (1963). "A digital computer for analyzing certain bioelectric signals," M.S. Thesis, Washington University, January.

- Davis, H., Engebretson, A.M., Lowell, E.L., Mast, T., Satterfield, J. and Yoshie, N. (1964). "Evoked responses to clicks recorded from the human scalp," *Annals of the New York Academy of Sciences*, 112 (1), 224-225, May 8.
- Miller, J.D., Engebretson, A.M. and Weston, P.B. (1964). "Recording the waveforms of periodic acoustic signals at levels near and below 0.0002 ubar," *J. Acoust. Soc. Am.* 36, 1951 (L).
- Engebretson, A.M. and Eldredge, D.H. (1968). "Model for the nonlinear characteristics of cochlear potentials," *J. Acoust. Soc. Am.* 44 (2), 548-554, August.
- Engebretson, A.M. (1969). "A study of the linear and nonlinear characteristics of microphonic voltage in the cochlea," D. Sc. dissertation, Washinton University, January.
- Fisher, W.M. and Engebretson, A.M. (1975). "Simple digital speech synthesis," *American Journal Computational Linguistics*, Microfiche 16.
- Miller J.D., Engebretson, A.M. and DeFilippo, C.L. (1976). "Preliminary research with a three-channel vibrotactile speech-reception aid for the deaf," In: G. Fant (Ed.) Speech Communication. Proceedings of the Speech Communication Seminar, Stockhom, April 1-3, 1974, New York: John Wiley & Sons, Vol. 4: Speech and Hearing, Defects and Aids, Language Acquisition, pp. 97-103.
- Monsen, R.B. and Engebretson, A.M. (1977). "Study of variations in the male and femal glottal wave," *J. Acoust. Soc. Am.* 62, 981-993.
- Monsen, R.B., Engebretson, A.M. and Vemula, N.R. (1978). "Indirect assessment of the contribution of subglottal air pressure and vocal-fold tension to changes of fundamental frequency in english," *J. Acoust. Soc. Am.* 64, 65-80, July.
- Vemula, N.R., Engebretson, A.M. and Elliott, D.L. (1979). "Models for the human throat-wall and a study of the vocal tract from input/output measurements," Proceedings of the Institute of Electrical and Electronic Engineers Conference on Decision and Control, San Diego, CA, pp. 946-948, January.
- Vemula, N.R., Engebretson, A.M., Monsen, R.B. and Lauter, J.L. (1979). "A speech microscope," In J.J. Wolff and D.H. Klatt (Eds.), Speech Communication Papers, New York: Acoustical Society of American, pp. 71-74, June.
- Monsen, R.B., Engebretson, A.M. and Vemula, N.R. (1979). "Some effects of deafness on the generation of voice," *J. Acoust. Soc. Am.* 66, 1680-1690, December.
- Miller, J.D., Engebretson, A.M. and DeFilippo, C.L. (1980). "Preliminary research with a three-channel vibrotactile speech-reception aid for the deaf," In H. Levitt, J.M. Pickett, and R.A. Houde (Eds.), Sensory Aids for the Hearing Impaired, New York: IEEE Press, pp. 341-347.
- Engebretson, A.M. and Miller, J.D. (1982). "A computer program for fitting a master hearing aid to the residual hearing characteristics of individual patients," *J. Acoust. Soc. Am.* 72 (2), 426-430, August.
- Monsen, R.B. and Engebretson, A.M. (1983). "The accuracy of formant frequency measurements: a comparison of spectrographic analysis and linear prediction," *J. Sp. and Hear. Res.*, Vol 26, 80-97.
- Popelka, G.R. and Engebretson, A.M. (1983). "A computer-based system for hearing aid assessment," *Hearing Instruments*, 34 (7), 6,7,9,4, July.
- Bade, P.R., Engebretson, A.M., Heidbreder, A.F. and Niemoeller, A.F. (1984). "Use of a personal computer to model the electroacoustics of hearing aids," *J. Acoust. Soc. Am.* 75: 617-620.

Engebretson, A.M. and O'Connell, M.P., "Implementation of a Microprocessor-based Tactile Hearing Prosthesis," submitted to IEEE Trans. Acoustics, Speech and Signal Processing, 1985.

ABSTRACTS AND CONFERENCE PAPERS:

- Engebretson, A.M. and Eldredge, D.H. (1967). "A model of the nonlinear behavior of cochlear microphonics and summing potential, J. Acoust. Soc. Am. 41, 1578 (A).
- Spenner, B.F., Engebretson, A.M., Miller, J.D. and Cox, J.R. (1973). "Random-access programmable recorder of complex sounds (RAP): A digital instrument for auditory research," Proceedings of the 86th Meeting of the Acoustical Society of America, October.
- Engebretson, A.M. and Vemula, N.R. (1974). "Study of the use of linear prediction and related methods of speech analysis for measuring vocal-tract area functions," presented at the 88th meeting of the Acoustical Society of America, St. Louis, Missouri, November.
- Miller, J.D., Engebretson, A.M. and DeFilippo, C.L. (1974). "Tactile speech-reception aids for the hearing impaired," presented at the 88th meeting of the Acoustical Society of America, St. Louis, Missouri, November.
- Monsen, R.B., Engebretson, A.M. and Fisher, W.M. (1974). "Some characteristics of the glottal sound source of deaf children," presented at the 88th meeting of the Acoustical Society of America, St. Louis, Missouri, November.
- Miller, J.D., Engebretson, A.M., Garfield, S.A. and Scott, B.L. (1975). "New approach to speech-reception testing," presented at the 89th meeting of the Acoustical Society of America, Austin, Texas, April, 1975. J. Acoust. Soc. Am. 57 (1), 548, Spring.
- Monsen, R.B., Engebretson, A.M. and Vemula, N.R. (1976). "Study of variations in the male and female glottal wave," presented at the 92nd meeting of the Acoustical Society of America, San Diego, California.
- Engebretson, A.M. (1977). "Computer system for auditory research (RAP-III)," presented at the 94th meeting of the Acoustical Society of America, Miami Beach, Florida.
- Miller, J.D., Engebretson, A.M., Spenner, B.F. and Cox, J.R. (1977). "Preliminary analysis of speech sounds with a digital model of the ear," presented at the 94th meeting of the Acoustical Society of America, Miami Beach, Florida.
- Miller, J.D., Engebretson, A.M. and Vemula, N.R. (1980). "Vowel normalization; differences between vowels spoken by children, women, and men," J. Acoust. Soc. Am. 68, Suppl. 1: 533 (A).
- Engebretson, A.M. and Tadlock, J.P. (1983). "A study of correlation methods for phoneme identification," J. Acoust. Soc. Am. 74, Suppl. 1: (A).
- Engebretson, A.M., Morley, R.E. and Popelka, G.R. (1983). "A unified digital hearing aid design and fitting procedure," American Speech-Language-Hearing Association, Cincinnati, Ohio.
- Popelka, G.R., Morley, R.E. and Engebretson, A.M. (1983). "Clinical advantages of a digital hearing aid," American Speech-Language-Hearing Association, Cincinnati, Ohio.
- Engebretson, A.M. (1983). "Statistics of short-term spectral characteristics of fluent speech," J. Acoust. Soc. Am. 73, Suppl. 1 (A).
- Engebretson, A.M. (1984). "Critical review of hearing-aid technology and future trends," invited paper to a conference sponsored by the Deafness Research Foundation, Washington, D.C., November 14.

- Engebretson, A.M. (1984). "A digital hearing aid," invited paper to XVII International Congress of Audiology, Santa Barbara, California, August 26-30.
- Engebretson, A.M. and O'Connell, M.P., "Implementation of a Real-Time, digital vocoder for tactile hearing prosthesis," Presented at International Conference on Acoustics, Speech, and Signal Processing, Tampa, Florida, March, 1985.
- Engebretson, A.M. and O'Connell, M.P., "Development of a microprocessor-based, tactile hearing prosthesis," to be presented at the 7th annual IEEE conference on Engineering in Medicine and Biology, Chicago, Illinois, September, 1985.

HONOR SOCIETIES:

- 1963 Sigma Xi
1963 Eta Kappa Nu

PROFESSIONAL SOCIETIES:

- 1972 American Institute of Electrical and Electronics Engineers
1981 Acoustical Society of America

REGISTERED PROFESSIONAL ENGINEER:

- 1981 State of Missouri

RESEARCH GRANTS AND CONTRACTS:

- 1983-1985 Principle Invest, Development of a Digital Hearing Aid and Fitting Procedure, Veterans Administration. (\$560,000)

CURRICULUM VITAE

PETER GAILLARD SMITH, M.D., Ph.D.

Date of Birth: January 15, 1945
Place of Birth: Ancon, Canal Zone
Citizenship: U.S.
Marital Status: Married (Constance Hallier)
Children: One (Todd Palmer)

EDUCATION

Clemson University, B.S., Chemical Engineering, 1967, Magna Cum Laude
Purdue University, M.S., Chemical Engineering, 1970, Magna Cum Laude
Purdue University, Ph.D., Chemical Engineering, 1972, Magna Cum Laude
Medical University of South Carolina, M.D., 1976, Magna Cum Laude

POSTGRADUATE TRAINING

Research Assistant, Dow Chemical Company, Walnut Creek, California, 1967
Graduate Research Fellow, Purdue University, Lafayette, Indiana, 1967-1972
Senior Research Engineer, Esso Production Research Company, Houston, Texas,
1972-1973
Resident in Surgery, Jewish Hospital, St. Louis, Missouri 1976
Resident and Fellow in Otolaryngology, Barnes Hospital and Washington
University School of Medicine, St. Louis, Missouri, 1977-1981
Fellow in Neurotology and Surgery of the Skull Base, The Otology Group,
Nashville, Tennessee, 1981

HOSPITAL APPOINTMENTS

Assistant Otolaryngologist, Barnes Hospital, St. Louis, MO, 1981-Present
Attending Staff, Jewish Hospital of St. Louis, St. Louis, MO 1981-Present
Attending Staff, Children's Hospital of St. Louis, St. Louis, MO 1981-Present
Consulting Staff, Veterans Administration Hospital, St. Louis, MO 1981-Present
Consulting Staff, Missouri Crippled Children's Division 1981-Present

ACADEMIC APPOINTMENTS

Assistant Professor, Department of Otolaryngology, Washington University School of Medicine, St. Louis, Missouri, 1982-Present.

Coursemaster, Basic Science of Otolaryngology, Washington University School of Medicine, St. Louis, Missouri, 1982-Present.

SOCIETY MEMBERSHIPS

American Institute of Chemical Engineers

Acoustical Society of America

American Medical Association

Missouri State Medical Association

Metropolitan Medical Society of St. Louis

American Academy of Otolaryngology - Head and Neck Surgery

Neurotologic Society (Fellow)

American College of Surgeons (Fellow)

Centurions of the Deafness Research Foundation

Missouri Society of Otolaryngology - Head and Neck Surgery

Barnes Hospital Society

Jewish Hospital Society

HONOR SOCIETIES

Alpha Omega Alpha

Sigma Xi

Tau Beta Pi

Phi Kappa Phi

AWARDS

President's Award, American Academy of Otolaryngology, Head and Neck Surgery, 1985

Special Teacher Award, Washington University, 1981

Resident Award, 1981

AWARDS (CONTINUED)

Lange Book Award, 1976, 1977

NSF Fellowship, 1970-1972

NASA Fellowship, 1967-1970

Dow Chemical Company Fellowship, 1970-1972

Dow Chemical Company Scholarship, 1964-1967

Monsanto Chemical Company Scholarship, 1964-1967

Gold Medal of Society of American Military Engineers, 1967

LICENSURES

Tennessee State Board of Medical Examiners, 1981

Missouri State Board of Healing Arts (R8720), 1978

MILITARY SERVICE

U.S.A. Corps of Engineers, Captain, Honorably Discharged, 1967-1975

COMMITTEES - LOCAL

Transfusion Committee, Barnes Hospital, St. Louis, MO 1981-Present

Transplantation Committee, Barnes Hospital, St. Louis, MO 1981-Present

Environmental and Safety Protection Committee, Barnes Hospital
St. Louis, MO 1980-Present

COMMITTEES - STATE

Credentials Committee - Missouri Society of Otolaryngology -
Head and Neck Surgery 1985

COMMITTEES - NATIONAL

American Academy of Otolaryngology Task Force for
Developing Educational Materials 1985

EDITORIAL BOARD MEMBERSHIP

The Laryngoscope 1981-Present

PUBLICATIONS

1. Smith PG: Frequency-dependent flow regimes in porous media. M.S. thesis, Purdue University, 1970.
2. Smith PG, Greenkorn RA: Sound wave propagation in porous media. Symposium on Mathematical Modeling of Transport Processes in Porous Media, AICHE National Meeting, 1971.
3. Smith PG: Transient response of rigid porous media. Ph.D. thesis, Purdue University, 1972.
4. Smith PG, Greenkorn RA: Theory of acoustical wave propagation in porous media. Journal of the Acoustical Society of America, 52:247-253, 1972.
5. Smith PG, Greenkorn RA, Barile RG: Infrasonic response characteristics of gas-and liquid-filled porous media. Journal of the Acoustical Society of America, 56:781-788, 1974.
6. Smith PG, Greenkorn RA, Barile RG: Theory of transient pressure response of fluid-filled porous media. Journal of the Acoustical Society of America, 56:789-795, 1974.
7. Spector GJ, Smith PG, Burde RM: Selective facial neurectomy for essential blepharospasm. The Laryngoscope, 91:1896-1903, 1981.
8. Smith PG, Thawley SE, Muntz HR: Local myocutaneous advancement flaps: alternatives to cross-lip and distant flaps in the reconstruction of ablative lip defects. Arch Otolaryngol, 108:676-681, 1982.
9. Thawley SE, Smith PG, Faw KD: The use of sclera in tympanic membrane reconstruction. The Laryngoscope, 92:1360-1362, 1982.
10. Marks JE, Lee F, Smith PG, Ogura JH: Floor of mouth cancer: a study of patient selection and treatment results. The Laryngoscope, 93:475-480, 1983.
11. Spector GJ, Smith PG: Some observations on endolymphatic sac surgery for Meniere's disease. Ann. Otol. Rhinol. Laryngol., 92:113-118, 1983.
12. Smith PG, Dyches TJ, Loomis RA: Clinical aspects of the branchio-oto-renal syndrome. To be published in Otolaryngol Head and Neck Surgery, 1984.
13. Jackson CG, Glasscock ME, Schwaber MK, Nissen AJ, Christensen SG, Smith PG: Ossicular chain reconstruction: The torp and porp in chronic ear disease. The Laryngoscope 93: 981-988, 1983
14. Muntz HR, Smith PG: Carotid artery hypersensitivity as a cause of syncope in patients with head and neck malignancies. The Laryngoscope, 93:1290-1293, 1983.
15. Glasscock ME, Smith PG, Whitaker SR, Bartels LJ: Management of aneurysms of the petrous portion of the internal carotid artery by resection and primary anastomosis. The Laryngoscope, 93:1445-1453, 1983.

PUBLICATIONS (CONTINUED)

16. Collins SL, Smith PG: Thin and skin-grafted pectoralis major myocutaneous flaps. Plastic and Reconstr Surg of the Head and Neck - Proc Fourth International Symp. P.H. Ward and W.E. Berman (Eds.), C.V. Mosby, Vol 2, Chapt. 139, pp 956-968, 1984.
17. Smith PG, Collins SL: Repair of head and neck defects with thin and double-lined pectoralis flaps. Arch Otolaryngol, 110:468-473, 1984.
18. Glasscock ME, Smith PG, Schwaber MK, Nissen AJ: Clinical aspects of hemangiomas of the skull base. The Laryngoscope, 94:869-873, 1984.
19. Schwaber MK, Nissen AJ, Smith PG, Jackson CG, Glasscock ME: Diagnosis and management of catecholamine secreting glomus tumors. Laryngoscope, 94:1008-1014, 1984.
20. Marks JE, Smith PG, Sessions DG: Pharyngeal wall cancer: a reappraisal after comparison of treatment methods. To be published in Arch Otolaryngol, 1985.
21. Smith PG, Dyches TJ: Topographical analysis of Horner's syndrome. To be published in Otolaryngol - Head and Neck Surgery, 1985.
22. Smith PG, Stroud MH, Goebel JA: Soft wall reconstruction of the posterior ear canal wall. To be published in Otolaryngol Head and Neck Surgery, 1985.
23. Matsuba HM, Thawley SE, Smith PG, Simpson J: Adenoid cystic carcinoma of major and minor salivary gland origin. The Laryngoscope (accepted for publication).
24. Matsuba HM, Thawley SE, Smith PG: Internal jugular vein phlebectasia. Head and Neck Surg (accepted for publication).
25. Matsuba HM, Thawley SE, Smith PG: Tension pneumocephalus. Am J Otolaryngology (accepted for publication).
26. Marks JE, Breaux S, Smith PG et al.: The need for elective irradiation of occult lymphatic metastases from cancers of the larynx and pyriform sinuses. Head and Neck Surgery (accepted for publication).

CHAPTERS

1. Smith PG, Schwaber MK, Goebel JA: Clinical evaluation of glomus tumors of the ear and skull base. To be published in Comprehensive Management of Head and Neck Tumors, SE Thawley, WR Panje (Eds.), W.B. Saunders, Philadelphia, 1984.
2. Smith PG, Lucente FA: Infectious diseases of the external ear. To be published in Otolaryngology - Head and Neck Surgery, CW Cummings, JM Fredrickson, LA Harker, CJ Krause, DE Schuller (Eds.), C.V. Mosby, St. Louis, 1984.

5. Lucence PL, Smith PG, Thomas OR. Diseases of the external ear. Otologic Medicine and Surgery. Alberti PW and Ruben RJ (Eds), Churchill Livingstone (in press).

PRESENTATIONS

1. Smith PG: Sensory-dependent flow regimes in porous media. Acoustical Society of America National Meeting, Houston, 1969.
2. Spector GJ, Smith PG, Burde RM: Selective facial neurectomy for essential blepharospasm. Triological Society, Middle Section Meeting, Oklahoma City, 1981.
3. Smith PG, Thawley SE, Muntz HR: Local myocutaneous flaps: alternatives to cross-lip and distant flaps in the reconstruction of ablative lip defects. American Academy of Otolaryngology - Head and Neck Surgery National Meeting, New Orleans, 1981.
4. Smith PG, Dyches TJ, Loomis RA: Clinical aspects of the branchio-oto-renal syndrome. American Academy of Otolaryngology - Head and Neck Surgery National Meeting, New Orleans, 1982.
5. Glasscock ME, Smith PG, Whitaker SR, Bartels LJ: Management of aneurysms of the petrous portion of the internal carotid artery by resection and primary anastomosis. Triological Society, Middle Section Meeting, St. Louis, 1983.
6. Muntz HR, Smith PG: Carotid artery hypersensitivity as a cause of syncope in patients with head and neck malignancies. Triological Society, Middle Section Meeting, St. Louis, 1983.
7. Collins SL, Smith PG: Thin pectoralis myocutaneous flaps. Fourth International Symposium on Plastic and Reconstructive Surgery of the Head and Neck, Los Angeles, 1983.
8. Smith PG, Collins SL: Repair of head and neck defects with thin and double-lined pectoralis flaps. American Academy of Otolaryngology - Head and Neck Surgery National Meeting, Anaheim, 1983.
9. Smith PG: Clinical aspects of hemangiomas of the skull base. Triological Society, Western Section Meeting, Santa Barbara, 1984.
10. Smith PG: Oncologic aspects of lateral skull base surgery. MSMA Annual Meeting, St. Louis, 1984.
11. Smith PG: Myogenous flap reconstruction of extensive head and neck defects. MSMA Annual Meeting, St. Louis, 1984.
12. Smith PG, Stroud MH, Goebel JA: Soft wall reconstruction of the posterior external canal wall. American Academy of Otolaryngology - Head and Neck Surgery National Meeting, Las Vegas, 1984.

PRESENTATIONS (CONTINUED)

13. Smith PG, Dyches TJ, Burde RM: Topographical analysis of Horner's syndrome. American Academy of Otolaryngology - Head and Neck Surgery National Meeting, Las Vegas, 1984.
14. Smith PG, Killeen TE: Management of arterial spasm complicating extensive skull base surgery. Annual meeting, Missouri Chapter, American College of Surgeons, 1984.
15. Smith PG, Glasscock ME, Matsuba H, Thawley SE: The infratemporal fossa approach to arterial and neoplastic lesions of the lateral skull base. American College of Surgeons Annual meeting, San Francisco, 1984.
16. Smith PG: Aural complications of otitis media. Otolaryngology Grand Rounds, Washington University School of Medicine, St. Louis, 1984.
17. Smith PG: Surgical management of glomus jugulare tumors, Neurosurgical Grand Rounds, Washington University School of Medicine, St. Louis, 1984.
18. Smith PG: Intracranial complications of otitis media. Otolaryngology Grand Rounds, Washington University School of Medicine, St. Louis, 1985.
19. Smith PG: Evaluation of the acoustic tumor suspect. Otolaryngology Grand Rounds, Washington University School of Medicine, St. Louis, 1985.
20. Smith PG: Diagnostic strategies in the evaluation of the acoustic neuroma suspect. ENT Club of St. Louis, St. Louis, 1985.
21. Smith PG: Diagnosis and management of paragangliomas of the head and neck. General Surgery Grand Rounds, Washington University School of Medicine, St. Louis, 1985.
22. Smith PG: Toward a cost-effective evaluation of the acoustic tumor suspect. MSMA Annual Meeting, Kansas City, 1985.
23. Smith PG: Surgical management of paragangliomas of the skull base, Otolaryngology Grand Rounds, Loyola University School of Medicine, Chicago, 1985.
24. Smith PG: Clinical aspects of neurotologic lesions of the skull base. Barnes Hospital Otolaryngology Update, St. Louis, 1985.
25. Smith PG: Diagnosis and management of paragangliomas of the skull base, Otolaryngology Grand Rounds, Medical University of South Carolina, 1985.
26. Smith PG, Killeen TK: A study of the topical control of arterial spasm in reconstructive surgery. AAFPRS Spring Meeting, Miami 1985.
27. Smith PG: Panel discussion: interstitial implantation. Annual Meeting American Endocurietherapy Society, St. Louis 1985

VISITING PROFESSORSHIP

Loyola University

1985

Medical University of South Carolina

1985

CURRICULUM VITAE
OF
JOHN MURRAY FREDRICKSON

B.A., (Science - U.B.C.), M.D., (U.B.C.)

F.R.C.S. (C), F.A.C.S.

Hon. Dr. Med. (Linkoping) Sweden

Hospital Address:

Washington University
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Home Address:

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St. Louis, MO 63105
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Name: John Murray Fredrickson
Born: March 24, 1931, Winnipeg, Manitoba
Citizenship: Canadian
Marital Status: Married: Alix (nee Gordon)
Children: Kristin, Lisa and Erik

DEGREES AND DIPLOMAS

B.A. University of British Columbia 1953
(Majors: Physics, Chemistry, Biology)
M.D. University of British Columbia 1957
(Thesis: "Calcium Homostasis"
Supervisor, Prof. H. Copp)
M.D. (Hon) Sweden 1975
(Cited for Excellence in Vestibular Research)

POSTGRADUATE AND RESEARCH TRAINING

Research in the Laboratory of Professor D.H. Copp, 1954-56
Department of Physiology, University of British Columbia, (Summers)
on Calcium Homostasis. (These experiments preceded the
discovery of "Calcitonin".)
Internship (Rotating) Vancouver General Hospital 1957-58
Resident (Pathology) Vancouver General Hospital 1958-59
Resident (Surgery and Medicine) Shaughnessy Hospital, 1959-60
Vancouver
Resident (Otolaryngology) University of Chicago 1960-63
Visiting Investigator, Department of Clinical 1963-65
Neurophysiology, University of Freiburg, Germany

CERTIFICATION

F.R.C.S. (Canada) Otolaryngology 1963
American Board of Otolaryngology 1966
F.A.C.S. 1968

LICENSURE

British Columbia	1957
California	1965
Ontario	1968
Missouri	1983

ACADEMIC APPOINTMENTS

Instructor in Surgery (Otolaryngology)
University of Chicago. Time spent as:

Visiting Investigator from the University of Chicago:
Research in Electrophysiology of the Central Vestibular
System with Professor H.H. Kornhuber, Department of
Clinical Neurophysiology, University of Freiburg, Germany 1963-65

Assistant Professor, Department of Surgery,
Division of Otolaryngology, Stanford University 1965-68

Associate Professor, Department of Otolaryngology,
University of Toronto, Toronto General Hospital 1968-77

Assistant Professor, Department of Physiology,
University of Toronto 1969-82

Faculty Member, Institute of Medical Science 1969-82

Cross Appointment in Speech Pathology 1975-82

Director of Clinical Sciences Division,
University of Toronto 1976-82

Professor, Department of Otolaryngology
University of Toronto 1977-82

Lindburg Professor and Head, Department of Otolaryngology,
Washington University School of Medicine 1982

PROFESSIONAL APPOINTMENTS

Assistant Professor, Division of Otolaryngology (Chief of
Head and Neck Surgery), Stanford University Medical Center 1965-68

Senior Otolaryngologist, Department of Otolaryngology,
Toronto General Hospital (Head of Vestibular Unit) 1968-82

Consultant, Princess Margaret Hospital, E.N.T. Cancer Clinic 1968-82

PROFESSIONAL APPOINTMENTS (CONTINUED)

Otolaryngologist-in-Chief, Barnes Hospital 1982
Otolaryngologist-in-Chief, St. Louis Children's Hospital 1982

HONORS AND AWARDS

General Proficiency Award, Faculty of Medicine, University of British Columbia 1955
Surgery Award, Faculty of Medicine University of British Columbia 1956
American College of Surgeons Medal for Movie: "Laryngeal and Pharyngeal Pouches" 1963
Research Award, American Academy of Ophthalmology and Otolaryngology for "Vestibular Disorders in Fourth Ventricle Lesions" 1964
Canadian Otolaryngological Society, Hodge Memorial Award for "Convergence and Interaction of Vestibular and Deep Somatic Afferents Upon Neurons in the Vestibular Nuclei of the Cat" 1965
University of Toronto Graham Campbell Prize for Worthy Contributions to Advance Sound Knowledge in Otolaryngology 1966
Honorary Medical Doctorate, Linkoping University, Sweden "In Recognition of Excellence in Vestibular Research" 1975
Award of Merit, American Academy of Ophthalmology and Otolaryngology. For Outstanding Contribution to the Academy's Educational Programs 1976

SOCIETIES

College of Physicians and Surgeons, British Columbia 1957
Canadian Medical Association 1958
Royal College of Physicians and Surgeons of Canada 1963
California College of Physicians and Surgeons 1965
American Board of Otolaryngology 1966
Pan Pacific Surgical Association 1966
Canadian Otolaryngological Society 1966
American Society for Head and Neck Surgery 1966

SOCIETIES (CONTINUED)

American Academy of Otolaryngology - Head and Neck Surgery (AAO-HNS)	1966
Barany Society	1967
Society of University Otolaryngologists (SUO)	1967
College of Physicians and Surgeons, Ontario	1968
Academy of Medicine, Toronto	1968
American College of Surgeons	1968
American Academy of Facial Plastic and Reconstructive Surgery	1973
American Laryngological Society	1977
American Broncho-Esophagological Association	1978
American Otological Society	1978
Collegium Oto-Rhino-Laryngologicum	1978
Society for Neurovascular Surgery	1984
Neurovascular Society of North America	1984

NATIONAL ACADEMIC ACTIVITIES

- Chairman of the Research Committee of the AAO-HNS	1983
- SUO Representative to the Association of American Medical Colleges (AAMC)	1983
- Guest site visitor for the NINCDS	1983

PATENTS

An Implantable Hearing Aid	1973
An Implantable Voice Box	1981

SELECTED ITEMS FROM BIBLIOGRAPHY

- Fredrickson JM, Griffith AW, Lindsay JR: Transverse Fracture of the Temporal Bone: A Clinical and Histopathological Study. Arch Otolaryng 78:770, 1963.
- Fredrickson JM, Schwarz D, Kornhuber HH: Konvergenz and Interaktion Vestibularer and Proprioceptive-somatosensibler Afferenzen an Neuronen der Vestibulariskerne der Katze. Pflugers Archiv fur die gesamte Physiologie, Band 281, 1964.
- Fredrickson JM, Figge U, Scheid P, Kornhuber HH: Vestibular Nerve Projection to the Cerebral Cortex of the Rhesus Monkey. Exper Brain Res 2:318, 1966.
- Fredrickson JM, Kornhuber HH, Goode RL: Nystagmus: Diagnostic Significance of Recent Observations. Arch Otolaryng 89:504-511, 1969.
- Mehler WF, Fredrickson JM: Interrelations of the Vestibular, Auditory, and Somatosensory Cortices in the Rhesus Monkey. Anatomical Records 166:408, 1970.
- Schwarz D, Fredrickson JM: Tactile Direction Sensitivity of Area 2 Oral Neurons in the Rhesus Cortex. Brain Research 27:397, 1971.
- Fredrickson JM, Pearson BW: Traumatic Ear Disorders Including Fractures of the Temporal Bone. In: Otolaryngology, Ed., G. English, 28, 1972.
- Odkvist LM, Rubin AM, Schwarz D, Fredrickson JM: Vestibular and Auditory Cortical Projection in the Guinea Pig. Exp Brain Res 18:279, 1973.
- Fredrickson JM, Tomlinson DR, Davis ER, Odkvist LM: Evaluation of an Electromagnetic Implantable Hearing Aid. Canadian J Otolaryng 2:53, 1973.
- Liedgren C, Odkvist LM, Fredrickson JM, Davis ER: The Effect of Marihuana on Hearing. Canadian J Otolaryng 5: #3, 233, 1976.
- Griffiths WV, Fredrickson JM, Bryce DP: An Implantable Electromagnetic Sound Source for Speech Production. Archives of Otolaryng 102:675, 1976.
- Debreceni AE, Fredrickson JM, Griffiths MV: An Implantable Electromagnetic Sound Source for Speech Production. Trans Am Soc Artif Intern Organs 23:2226, 1977.
- Hawrylshyn PA, Rubin AM, Tasker RR, Organ LW, Fredrickson JM: Vestibulothalamic Projections in Man -- A Sixth Primary Sensory Pathway. J Neuro 41:394, 1978.
- Charles DA, Fredrickson JM, Bryce DP: The Electro-Magnetic Implantable Sound Source Preliminary Results of Human Implantation. Chapter In: Surgical and Prosthetic Approaches to Speech Rehabilitation. Eds, D.P. Shedd and B. Weinberg, 1980.
- Malmgren LT, Berlin CI, Fredrickson JM, et al: First National Conference on Research Goals and Methods in Otolaryngology. 5. Postdoctoral Research Training. Annals of Otolaryngology, Rhinology & Laryngology Supplement 100, Vol. 91, Nov-Dec 1982, No. 6, Part 3.

SUSAN BINZER

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St. Louis, Missouri 63104
314-241-4530

EDUCATION: M.A. Audiology, University of Cincinnati, 1984.
B.S. Speech Pathology and Audiology, University of Cincinnati,
1981.

EDUCATIONAL
EXPERIENCE:

Washington University Medical School, Department of
Otolaryngology, August 1984 - present.
Coordinator of Cochlear Implant and Aural Rehabilitation Programs.

James Whitcomb Riley Hospital for Children, Indianapolis,
Indiana, August 1983 - May 1984.
Evaluated auditory disorders of patients of all ages and abilities
and provided rehabilitative services to same. Administered and
interpreted BSER's. Assisted with assessment and rehabilitation of
cochlear implant patients.

Cincinnati Speech and Hearing Center, January 1983 - July 1983.
Provided evaluation and rehabilitative services for a variety of
auditory disorders.

Saint Rita School for the Deaf and Hard-of-Hearing, Cincinnati,
Ohio, October 1981 - June 1983.
Provided a variety of audiological services, including testing,
troubleshooting hearing aids, and making ear impressions. Provided
evaluation and remediation of speech, communication and language
disorders.

University of Cincinnati Summer Hearing Therapy Program, July
1982.
In conjunction with graduate speech pathology student, provided
intensive speech and language therapy for severely hearing-impaired
girl.

Cincinnati General Hospital (now University Hospital), June -
December 1982.
Evaluated auditory disorders of patients in ENT Clinic, school clinic
program, and private patients. Participated in rehabilitation of
private patients. Administered and interpreted ENG's.

Saint Elizabeth's Hospital, North Unit, Covington, Kentucky,
January - June 1982.
Performed basic audiological testing on new hospital employees.

Withamsville-Tobasco Elementary School
Merwin Elementary School
Amelia High School, Cincinnati, Ohio, March - June 1981.
Assisted in evaluation and remediation of a variety of
communicative disorders.

Arlitt Preschool, Cincinnati, Ohio, January - March 1981.
Participated in basic care of teenagers and provided individual attending therapy.

Steppingstones Center for the Handicapped, Cincinnati, Ohio,
September - December 1980.
Participated in basic care of teenagers and provided individual attending therapy.

WORK

EXPERIENCE:

University of Cincinnati Central Library, September 1981 -
September 1982.

Performed a variety of duties for the reference department.

Public Library of Cincinnati and Hamilton County, March 1977 -
August 1980, June - September 1982.

Responsibilities ranged from circulation of books to opening
and closing the library.

Kenwood Developmental Center for Brain-Injured Children,
Cincinnati, Ohio, Summer 1974.

Fed and exercised brain-injured children of all ages.

SCHOLARSHIPS/

AWARDS:

Graduate Teaching Assistantship, 1983.

University Graduate Scholarship, 1982, 1983.

Alpha Sigma Psi Honor Sorority

Dean's List for 10 quarters; graduated magna cum laude.

ASSOCIATIONS:

Member of American Speech and Hearing Association.

Member of A.G. Bell Association for the Deaf.

Member of Speech and Hearing Association of Greater St. Louis.

Member of National Student Speech Language and Hearing
Association, 1982-1984.

Member of Greater Indianapolis Speech and Hearing Association,
1983-1984.

Member of Southwestern Ohio Speech and Hearing Association,
1980-1983.

President of Speech Pathology/Audiology division of the
Graduate Student Association, 1982-1983.

President of the University of Cincinnati chapter of the
National Student Speech Language and Hearing Association,
1982-1983.

PAPERS:

"Satisfaction of hearing aid users when comparing closeness
of fit of their aids to a computer recommendation." M.A.
thesis. Presented at Academy of Rehabilitative Audiology, Summer
Institute, 1983 and Ohio Speech and Hearing Association Convention,
1984.

V. Statement of Work, Schedule and Budget

Independently and not as an agent of the Government, Research Triangle Institute (RTI) will exert its best efforts to design and develop speech processors for use with auditory prostheses. The Description of Work as given in RFP No. NIH-NINCDS-85-09 is acceptable to RTI, and is reproduced below for reference.

Specifically, RTI will:

- A. Design and develop a computer-based, multichannel waveform generator which when coupled with the collaborating investigators' multichannel neural stimulators will permit studies on:
 - 1. Improved temporal and spatial resolution of stimuli.
 - 2. The extension of the dynamic range of intensity coding.
 - 3. The extension of the range of frequency coding.
 - 4. Methods of preserving the subjective sensation of constant pitch while changing the subjective sensation of loudness.
 - 5. The factors which control the subjective sensation of pitch.

- B. Design and develop a computer-based, multichannel auditory signal processor for use in evaluating promising speech extraction and stimulus encoding schemes.
 - 1. The processor shall contain microphones to capture speech signals under various environmental conditions and convert them to electrical signals for preprocessing.
 - 2. Sufficient preprocessing circuitry shall be included to compensate for known psychophysical limitations of present multichannel auditory prostheses such as limited dynamic range of input

signals as determined by subjective loudness and limited rate pitch discrimination.

3. Devise hypotheses of potentially feasible speech processing schemes based on presently known psychophysical data from auditory prosthesis implant patients and develop software for testing them using a laboratory-based computer such that:
 - a. Their key elements can be varied independently.
 - b. They can be evaluated in human subjects in conjunction with current designs of single electrode or multiple electrode auditory prostheses.
 - c. They can be used to further evaluate multielectrode psychophysical characteristics including complex interelectrode interactions.
 - d. Comparisons can be made between different speech processing schemes in the same implant patient.
 - e. The essence of each of the speech processing schemes could be reduced to a real time, hardware based, wearable speech processor.
- C. Design and fabricate wearable speech processors based on the results obtained with the computer-based simulated designs such that:
1. They are designed for specific patients with single or multielectrode auditory prostheses.
 2. They are human engineered with respect to weight, durability, and panel component selection and placement.
 3. They take advantage of the implanted electrode configurations.
 4. They operate in real time.
 5. They can be used for studying the long-term effects of learning.

D. Supply at least two of these wearable speech processors to the Project Officer by at least three years after the start of the contract.

1. Include details of suggested procedures for evaluating these speech processors.

E. Assist the collaborating human subject evaluation team in implementing the above mentioned waveform generator, computer-based signal processor and the wearable speech processors.

Our general plan for the first year of work in this "expanded scope" contract includes the following:

1. Continue work on software to generate the stimuli for studies on "stimulus primitives";
2. Complete work on the block-diagram compiler, to improve the user interface and to incorporate the additional modules required for simulation of all processing strategies outlined in sections III.B and III.C;
3. Automate procedures to obtain measures of psychophysical performance and speech understanding;
4. Collaborate closely with the teams at UCSF, DUMC, and Washington University in the design and support of experiments to evaluate stimulus primitives and speech processing strategies;
5. Build and install a hardware interface for use at Washington University to allow communications between their Eclipse computer and implanted electrodes;

6. Assist the St. Louis team in configuring a laboratory for testing of cochlear implant patients;
7. Continue development of 80C31-based processors, including implementation of processors that have dual 80C31s or one 80C31 and a multiplier chip;
8. Define design changes that would be required to implement strategies outlined in section III.C of this proposal using our 80C88-based processors;
9. Evaluate alternative hardware designs for speech processors for auditory prostheses; and
10. Prepare and submit reports of progress at the end of each quarter, as specified in the RFP.

Work in years 2 and 3 would be mainly directed at completion of tasks 1, 3, 4, 7, and 8 above, and at design and construction of hardware processors based on the most promising results obtained in the evaluation studies.

The budget requested for this "expanded scope" contract is substantially larger than the budget for our present contract. This increase will allow the RTI team to (1) increase its effort in supporting the teams at UCSF and DUMC, to evaluate a broad range of processing strategies and stimulus primitives; (2) help initiate and continue to support a new evaluation effort at Washington University and Central Institute for the Deaf; (3) thoroughly evaluate more alternative designs for portable, real-time processors than the necessarily limited number described in the "present scope" proposal; and (4) provide additional take-home processors for patient use and for further evaluation of possible learning effects.

VI. Protection of Human Subjects

All patient testing associated with this proposed project will be performed at the collaborating institutions named in section IV.C of this proposal; however, RTI also provides its assurance of protection of human subjects and presents on the next page a completed Form HHS-596, "Protection of Human Subjects." Approved IRB forms from the University of California at San Francisco, Duke University Medical Center, and/or Washington University (pending) will be supplied on request.

VII. RTI Experience

As the incumbent for the project described in the present RFP, RTI has direct experience for the work outlined in this proposal. In particular, we have developed many of the tools to be used in the next project (see sections II.A-C); have direct experience from our current project and other projects at RTI in the design and construction of portable, microprocessor-based instruments that execute ~~in~~ real-time algorithms of the complexity required for the present applications (see sections II.E and III.E); and have designed stimulus primitives and speech-coding strategies that have clear promise for improving the performance of auditory prostheses (see sections III.A-C).

We believe, and hope the reviewers of this proposal will agree, that we have come a long way in 18 months. Not only have we accomplished the work briefly indicated above, but have been able to (1) build a powerful tool for understanding and defining the "electrical-to-neural transformer" linking the outputs of the speech processor to the inputs of the central nervous system and (2) help initiate a parallel testing effort at Duke. In addition, we have ~~made~~^{made}, or soon will make, the presentations listed below. Most of these presentations were invited and members of our team have been asked to chair sessions on cochlear implants (also listed below). These invitations are a gratifying measure of the wide recognition our work is receiving. Finally, in addition to the three full-length papers from our group that will be published in the proceedings of the various conferences listed below, we have several manuscripts in preparation reporting aspects of the work reviewed in this proposal. These papers include ones on coding strategies for auditory prostheses (sections III.A-C); the integrated field-neuron model (section II.B); and the computer-based simulator of speech processors for auditory prostheses (section II.C).

Lastly, RTI now has four projects directly related to the development of auditory prostheses; in addition, RTI has extensive experience in other fields relevant to the work outlined in the present RFP. A list of these projects is presented in Table VII.1 and selected abstracts of other RTI projects are presented at the end of this subsection to indicate our experience for the work at hand.

Record of reporting activity for the first 18 months of NIH project

N01-NS-2356, "Speech Processors for Auditory Prostheses"

Wilson, BS and CC Finley: Speech processors for auditory prostheses. To be presented at the International Cochlear Implant Symposium and Workshop, Melbourne, Australia, Aug. 27-31, 1985 (full-length paper to be published in the proceedings).

Finley, CC and BS Wilson: Field models of the Melbourne electrode array. Invited paper to be presented at the International Cochlear Implant Symposium and Workshop, Melbourne, Australia, Aug. 27-31, 1985 (full-length paper to be published in the proceedings).

Wilson, BS: Coding strategies for multichannel auditory prostheses. Invited paper to be 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 to be 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 to be 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 to be presented in the special session on signal processing for the hearing impaired, IEEE Bioengineering Conf., Sept. 27-30, 1985 (full-length paper to be published in the proceedings).

Wilson, BS and CC Finley: A computer-based simulator of speech processors for auditory prostheses. ARO Abstracts, 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. ARO Abstracts, 8th Midwinter Research Conference, p. 105, 1985.

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

Table VII.1. Present projects at RTI directly related to the development of cochlear implants

<u>PI</u>	<u>Title</u>	<u>Funding Source</u>
B. Wilson	"Speech Processors for Auditory Prostheses"	NIH
B. Wilson	"Evaluate the Efficacy of Single-Channel Coding Strategies for Extracochlear Auditory Prostheses"	Storz Instrument Company
C. Finley	"Prepare and Present an Invited Paper at the <u>1985 International Cochlear Implant Symposium</u> in Melbourne, Australia"	RTI Professional Development Award
B. Wilson	"Center for the Severely Hearing Impaired"*	Duke Surgery and RTI

*Initial tasks for this project are to construct a cochlear-implant laboratory at Duke Medical Center and otherwise help to establish an active program for development and clinical application of auditory prostheses at Duke.

DEVELOPMENT OF A SPEECH AUTOCUER

RTI Project No. 42U-1878

Sponsor: NASA Goddard Space Flight Center
(Contract No. NAS5-25832)

The objective of the autocuer project is to develop and field test a wearable speech-analyzing lipreading aid for deaf people. Based on the principles of Cued Speech, the autocuer must perform a phoneme-like speech analysis in real time of sufficient quality to disambiguate lipreading. The output of the speech analysis is a CV-syllable cue presented visually. These syllable cues in combination with lipreading are designed to enable clear perception of speech through vision alone.

In previous research, using simulation of automatically cued speech via videotapes, we demonstrated 84% cued word recognition accuracy versus 61% uncued accuracy (5 deaf subjects, 1 hearing) on a trained vocabulary of 500 words. Related research recently completed by Nicholls and Ling with manually cued speech has shown near-perfect reception of cued sentences by profoundly deaf subjects (96% accuracy of key words in sentences, 18 subjects, 97 dB or worse pure tone average audiogram).

The autocuer project is a four-way collaboration between Research Triangle Institute (speech analysis, project coordination) Gallaudet College (laboratory training and testing, field test) NASA Goddard Space Flight Center (CMOS LSI speech preprocessors/parameter extractors), and Telesensory Systems, Inc. (fabrication of field test units).

To be five years in duration, the project began in August 1979 and is funded by NASA and the VA.

SPEECH PROCESSORS FOR AUDITORY PROSTHESES

RTI Project No. 425U2727

Sponsor: National Institutes of Health
(Contract No. N01-NS-2356)

The purpose of this project is to design and evaluate speech processors for multichannel auditory prostheses. Ideally, the processors will extract (or preserve) from conversational speech those parameters that are essential for intelligibility and then appropriately encode these parameters for electrical stimulation of the auditory nerve on a sector-by-sector basis. Major tasks in our project include the following: (1) identify and contrast the most promising approaches to the design of speech processors for multichannel auditory prostheses; (2) build a computer-based simulator that is capable of rapid and practical emulation of all these approaches in software; (3) design and fabricate a hardware interface that will provide a communications link between the computer and implanted electrodes; and (4) evaluate promising strategies for speech processing in tests with single subjects so that meaningful comparisons of performance can be made. The tests of task 4 are being conducted in collaboration with investigators at the University of California at San Francisco (UCSF). Our colleagues at UCSF are also actively involved in the work of tasks 1, 2 and 3. Finally, arrangements have been made to conduct parallel tests at the Duke University Medical Center using procedures identical to those used in the tests at UCSF.

DESIGN, BUILD AND TEST AN AUDITORY-NERVE SIMULATOR

RTI Project No. 42U-9567

Sponsor: National Institutes of Environmental Health Sciences
(Contract No. PR-04821)

The objective of this project is to provide an analog instrument which faithfully mimics the neural encoding of sound at the auditory periphery. Features of signal transformation and transmission from the excitation of cochlear hair cells to the propagation of action potentials in VIII-th nerve axons will be modeled using the circuit designed by Evans* in 1979 for this purpose. Tuning properties inherent in the traveling wave of the basilar membrane will be modeled with a bandpass filter, with skirts of 50 dB/octave. The contractor will (1) design the filter, (2) design offset-compensation networks for the Evans simulator; (3) interface the filter to the simulator; (4) build an instrument with these components and with an appropriate power supply; (5) align the instrument according to the procedure specified by Evans**; and (6) evaluate the instrument's performance in terms of its ability to reproduce the primary characteristics of auditory-nerve function as reflected in discharge rate-intensity curves, tuning curves, and poststimulus-time histograms.

* E. F. Evans, "An Electronic Analogue of Single Unit Recording from the Cochlear Nerve for Teaching and Research, "J. Physiol. (Lond.), 298: 6-7P, 1979.

**E. F. Evans in a personal communication to B. Wilson of the Research Triangle Institute, RTP, NC.

INVESTIGATIONS TO DETERMINE THE PERIPHERAL AND CENTRAL RECEPTORS
MEDIATING EFFECTS OF MICROWAVE RADIATION ON BRAIN ACTIVITY

RTI Project No. 42U-1827

Sponsor: National Institute of Environmental Health Sciences, RTP
(Contract No. N01-ES-9-0008)

Recent findings indicate that brain activity is altered during exposures of animals to nonionizing radiation at average power densities of 10 mW/sq. cm or less. Among these findings are perception of pulsed microwave energy as auditory sensations and effects of microwave, very high frequency (VHF), and extremely low frequency (ELF) radiation on the electroencephalogram and behavior.

The main objective of this research is to identify the sites at which nonionizing radiation acts in eliciting auditory responses. Recording and subsequent analysis of single-unit activity in the cat's auditory nerve will be used to compare the latencies and patterns of responses to microwave pulses and acoustic clicks. Selected results obtained from these comparisons will be corroborated in the rat using the [^{14}C]-2-deoxy-D-glucose ([^{14}C]2DG) method. Finally, the [^{14}C]2DG method will also be used to evaluate the possibilities of microwave-evoked responses in the vestibular system and hypothalamus.

IDENTIFICATION OF SITES IN BRAIN TISSUE AFFECTED
BY NONIONIZING EMR

RTI Project No. 42U-1903

Sponsor: Environmental Protection Agency (Contract No. 68-02-3276)

Recent findings indicate that brain activity is altered during exposures of animals to nonionizing radiation at average power densities of 10 mW/sq. cm or less. The main objective of this study is to identify the sites at which nonionizing radiation acts in eliciting these responses. Test animals will be separately exposed to pulsed and continuous-wave 918 MHz radiation at 80, 10, 5 and 0 mW/sq. cm. Possible differences in brain activity between control (sham irradiated) and exposed animals will be measured using [^{14}C]-2-deoxy-D-glucose. This method has an enormous sampling advantage over classical electrophysiological methods in that it allows for simultaneous determination of glucose utilization and associated functional activity in most macroscopic structures of the brain. Thus, we expect that patterns of [^{14}C]2DG uptake in the brains of exposed animals might reveal alterations in activity at sensory nuclei, resulting from stimulation of peripheral receptors, as well as other possible effects on brain activity, resulting from radiation-induced changes in the neural environment.

DESIGN AND FABRICATE WEARABLE CARDIOPULMONARY PERSONAL MONITOR

RTI Project No. 42U-2484

Sponsor: Environmental Protection Agency (Contract No. 68-02-3774)

This contract is for the design, development, clinical validation, and limited field test of a wearable medical instrument for continuous, non-invasive collection of cardiopulmonary data. The subject instrument, called the cardiopulmonary personal monitor (CPM), is being designed to enable the assessment of cardiac changes in normal and risk subjects resulting from exposure to cardiotoxins such as carbon monoxide, ozone, diesel exhaust pollutants, and those that might be present at toxic dump sites. The data collected by the CPM will be downloaded to and analyzed by a laboratory-based microcomputer system. The CPM will be designed to acquire data which will allow derivation of the following information

- Left Ventricular Ejection Time (LVET)
- Pre-Ejection Period (PEP)
- Heart Rate (HR)
- PEP/LVET ratio
- Isovolumic Contraction Time
- Estimated Left Ventricular Ejection Fraction
- Estimated Left Ventricular Filling Pressure
- Estimated Pulmonary Capillary Wedge Pressure

at heart rates up to 150 beats per minute. Project activities include hardware and software design, prototype fabrication, clinical trials, and field test demonstration of the developed cardiopulmonary personal monitor.

BIOMEDICAL ENGINEERING SUPPORT

RTI Project No. 42U-1534

Sponsor: Environmental Protection Agency

Research Triangle Institute (RTI) provided a variety of biomedical engineering support functions to the Human Studies Division of the Environmental Protection Agency on a task order basis. The scope of work performed includes electronic systems development, microcomputer software development, coinvestigation with EPA human studies, statistical data analysis, and feasibility design of new biomedical instrumentation. The principal activity for each task was:

- Task 1 - Microcomputer software development for the calculation of cardiac ventricular volumes and systolic time intervals from an echocardiogram.
- Task 2 - A feasibility study on the use of impedance cardiography for the measurement of systolic time intervals.
- Task 3 - Perform analog preprocessing modifications on an automatic ultrasound range-gate tracking system to improve overall performance.
- Task 4 - Develop a cardiac exercise stress processor (CESP) for automated electrocardiogram and impedance cardiogram processing during exercise stress testing.
- Tasks 5&6-A feasibility study and preliminary development of personal (wearable) cardiac monitor devices for EPA epidemiologic studies.
- Tasks 7&8-Fabricate two additional CESP systems for EPA clinical studies including various software enhancements.
- Task 9 - Investigate various methods of noninvasive cardiac function monitoring, design appropriate instrumentation, and validate the methods in joint EPA-RTI human studies.
- Task 10 - Design and fabricate a wearable electrocardiograph and impedance cardiograph for EPA exposure chamber studies.

ELECTRICAL SAFETY MAINTENANCE AND SERVICE OF
BIOMEDICAL RESEARCH EQUIPMENT

RTI Project Nos. 42U-1375 and 42U-2384

Sponsor: Environmental Protection Agency (Contract Nos. 68-02-2455
and 68-02-3753)

These contracts established and continue to carry out a systematic electrical safety maintenance and service program within the Human Studies Division (HSD) of the Health Effects Research Laboratory of the United States Environmental Protection Agency, located in Chapel Hill, North Carolina. The program insures that an electrically safe clinical research environment exists in the HSD laboratories for human test experiments. A wide variety of biomedical instrumentation which interfaces with test subjects and researchers, and associated research and support equipment, must be inspected, maintained, and periodically certified as electrically safe.

In support of an effective electrical safety program, a set of operational guidelines for electrical safety within HSD is required. The adopted guidelines are based on recommendations and standards promulgated by various standards groups, including the National Fire Protection Association, the Association for the Advancement of Medical Instrumentation, Underwriters Laboratories and the Joint Commission on Accreditation of Hospitals. Electrical safety manuals, produced under the contracts, incorporate these guidelines and other pertinent electrical safety information. The manuals are intended to be educational and reference documents with which the electrical safety of all HSD experimental protocols can be verified. The manuals are also the basic references specifying the procedures and guidelines to which the operation of the electrical safety control program must conform for all electrical safety inspection, certification, and maintenance activities. RTI monitors the development of new/revised standards and operational guidelines pertaining to electrical safety in the clinical environment and periodically presents training seminars to HSD personnel. As warranted, the HSD electrical safety program is modified to conform to revised standards and guidelines.

BIOMEDICAL APPLICATIONS TEAM

RTI Project No. 42U-2016

Sponsor: National Aeronautics Space Administration
(Contract No. NAS1-16177)

Under the sponsorship of the National Aeronautics and Space Administration, the Research Triangle Institute has operated a Biomedical Applications Team for 16 years. The objectives of this program are to transfer aerospace technology to applications in medicine and to achieve widespread availability and utilization of these applications within the medical community. These objectives are best accomplished by the development of commercially available medical products that incorporate aerospace technology. The RTI team's activities in meeting these objectives may be divided into four phases: (1) identifying medical applications of NASA technology, (2) screening applications to identify those that represent potentially successful commercial products, (3) developing commercialization strategies that include necessary adaptation of NASA technology, development funding, patent status and FDA approval, and (4) implementing and monitoring commercialization strategies.

Some of the devices currently under development and evaluation in this program are described below.

<u>Device</u>	<u>NASA Technology</u>	<u>Manufacturer</u>
Programmable Implantable Medication Delivery System	Fluid handling technology and telemetry	Pacesetter Systems
Implantable Prosthetic Urinary Sphincter	Miniaturized, high reliability valves	Medical Engineering Corp. & Parker Hannifin
Lightweight Wheelchair	Composite materials	Inductron Corporation

CLINICAL VALIDATION AND SYSTEMS DESIGN FOR CARDIOPULMONARY
ASSESSMENT OF CARBON MONOXIDE EXPOSURES

RTI Project No. 42U-2429

Sponsor: Environmental Protection Agency (Contract No. 68-02-3765)

RTI as a coinvestigator with EPA and University of North Carolina scientists will participate in the clinical validation of physiological models which estimate blood carboxyhemoglobin depending upon ambient carbon monoxide concentrations, ventilatory rate, work level, and other factors. RTI's participation will involve (1) mathematical analysis and modeling, (2) defining experimental protocols, conducting experiments and reporting results, and (3) developing and modifying special instrumentation as needed to achieve program objectives. This project consists of a variety of biomedical, electronic, and software engineering tasks and responsibilities and may be combined into four broad categories:

- Review and analysis of the Coburn-Forster-Kane equation.
- Modification of existing Cardiac Exercise Stress Processor (CESP) hardware and software resources.
- Development of new CESP software for database management, CFK modeling, statistical analysis, and graphic presentation.
- Research in impedance electrode technology.

DEVELOPMENT AND CLINICAL VALIDATION OF CARDIAC LEFT
VENTRICULAR FILLING PRESSURE AND SYSTOLIC TIME INTERVAL
MEASUREMENTS USING IMPEDANCE CARDIOGRAPHY

RTI Project No. 42U-2538

Sponsor: Duke University (subcontract to EPA No. CR810264-01-0)

Research Triangle Institute and Duke University investigators will study impedance cardiographic estimates of LV valvular events and left atrial pressure (LAP) in patients undergoing simultaneous diagnostic Swan-Ganz catheterization and M-mode echocardiographic examination. The goals of this study will be (1) comparison of systolic time intervals (STI) derived from echocardiography and the first derivative of the impedance cardiogram, (2) comparison of LAP estimates from echocardiography, impedance cardiography, and Swan-Ganz catheterization, and (3) clinical validation of the EPA-HERL impedance cardiographic system. Contingent upon favorable results of the clinical studies, a system for simultaneous STI and LAP measurements using the noninvasive impedance cardiogram will be developed.

References mistakenly left out of the original proposals

- Allen, J. B., Magnitude and phase-frequency response to single tones in the auditory nerve, J. Acoust. Soc. Am., 73 (1983) 2071-2092.
- Black, R. C. and Clark, G. M., Differential electrical excitation of the auditory nerve, J. Acoust. Soc. Am., 67 (1980) 868-874.
- Goldstein, J. L. and Sruлович, P., Auditory-nerve spike intervals as an adequate basis of aural frequency measurements. In E. F. Evans and J. P. Wilson (Eds.), Psychophysics and Physiology of Hearing, Academic Press, London, 1977, pp. 337-346.
- Johnson, D. H. and Kiang, N. Y. S., Analysis of discharges recorded simultaneously from pairs of auditory nerve fibers, Biophysical J., 16 (1976) 719-734.
- Loeb, G. E., White, M. W. and Merzenich, M. M., Spatial cross-correlation: A proposed mechanism for acoustic pitch perception, Biol. Cybernetics, 47 (1983) 149-163.
- Sachs, M. B., Speech encoding in the auditory nerve. In C. Berlin (Ed.), Hearing Science, College-Hill Press, San Diego, 1984, pp. 263-307.
- Sinex, D. G. and Geisler, C. D., Comparison of the responses of auditory nerve fibers to consonant-vowel syllables with predictions from linear models, J. Acoust. Soc. Am., 76 (1984) 116-121.
- Sruлович, P. and Goldstein, J. L., A central spectrum model: A synthesis of auditory-nerve timing and place cues in monaural communication of the frequency spectrum, J. Acoust. Soc. Am., 73 (1983) 1266-1276.
- Stypulkowski, P. H. and van den Honert, C., Physiological properties of the electrically stimulated auditory nerve. I. Compound action potential recordings, Hearing Res., 14 (1984) 205-223.
- van den Honert, C. and Stypulkowski, P. H., Physiological properties of the electrically stimulated auditory nerve. II. Single fiber recordings, Hearing Res., 14 (1984) 225-243.
- Wakefield, G. H. and Nelson, D. A., Extension of a temporal model of frequency discrimination: Intensity effects in normal and hearing-impaired listeners, J. Acoust. Soc. Am., 77 (1985) 613-619.
- White, M. W., Merzenich, M. M. and Gardi, J. N., Multichannel cochlear implants, Arch. Otolaryngol., 110 (1984) 493-501.

VIII. References

- Atal, B. S., Speech coding: Recognizing what we do not hear in speech, Ann. N. Y. Acad. Sci., 405 (1983) 18-32.
- Bilger, R. C., Evaluation of subjects presently fitted with implanted auditory prostheses, Ann. Otol. Rhinol. Laryngol., 86, Suppl. 38 (1977) 1-176.
- Bilger, R. C., Auditory results with single-channel implants, Ann. N. Y. Acad. Sci., 405 (1983) 337-342.
- Chouard, C. H. and MacLeod, P., Implantation of multiple intracochlear electrodes for rehabilitation of total deafness: preliminary report, Laryngoscope, 86 (1976) 1743-1751.
- Clark, G. M. and Tong, Y. T., A multiple-channel cochlear implant: A summary of results for two patients, Arch. Otolaryngol., 108 (1982) 214-217.
- Clark, G. M., Tong, Y. C., Martin, L. F. A. and Busby, P. A., A multiple-channel cochlear implant, Acta Oto-laryngol., 91 (1981) 173-175.
- Diller, N., Spillmann, T., Fisch, U. P. and Leifer, L. J., Encoding and decoding of auditory signals in relation to human speech and its application to human cochlear implants, Audiology, 19 (1980) 146-163.
- Dobie, R. and Dillier, N., Some aspects of temporal coding for single-channel electrical stimulation of the cochlea. Paper presented at the West Coast Cochlear Prosthesis Meeting, Seattle, 1984.
- Dowell, R. C., Brown, A. M., Blamey, P. J., Clark, G. M. and Seligman, P. M., Evaluation of a two-formant speech processing strategy for the Nucleus multi-channel cochlear prosthesis. In ARO Abstracts, 8th Midwinter Research Conference, p. 110, 1985.

- Eddington, D. K., Speech discrimination in deaf subjects with cochlear implants, J. Acoust. Soc. Am., 68 (1980) 885-891.
- Eddington, D. K., Dobelle, W. H., Brackmann, D. E., Mladejovsky, M. G. and Parkin, J. L., Auditory prosthesis research with multiple channel intracochlear stimulation in man, Ann. Otol. Rhinol. Laryngol., 87, Suppl. 53 (1978) 1-39.
- Evans, E. F., Place and time coding of frequency in the peripheral auditory system: Some physiological pros and cons, Audiology, 17 (1978) 369-420.
- Flanagan, J. L. and Golden, R. M., Phase vocoder, Bell Syst. Tech. J., 45 (1966) 1494-1509.
- Flanagan, J. L., Speech Analysis, Synthesis and Perception, Springer-Verlag, Berlin, 1972, 444 pp.
- Fourcin, A. J., Rosen, S. M., Moore, B. C. J., Douek, E. E., Clarke, G. P., Dodson, H. and Bannister, L. H., External electrical stimulation of the cochlea: Clinical, psychophysical, speech-perceptual and histological findings, Brit. J. Audiol., 13 (1979) 85-107.
- Golden, R. M., Digital computer simulation of sampled-data communication systems using the block diagram compiler: BLODIB, Bell Syst. Tech. J., 45 (1966) 345-358.
- Hinojosa, R. and Marion, M., Histopathology of profound sensorineural deafness, Ann. N. Y. Acad. Sci., 405 (1983) 459-484.
- Hochmair, E. S. and Hochmair-Desoyer, I. J., Aspects of sound signal processing using the Vienna intra- and extracochlear implants. In R. A. Schindler and M. M. Merzenich (Eds.), Cochlear Implants, Raven Press, New York, 1985, pp. 101-110.
- Hochmair, E. S., Hochmair-Desoyer, I. J. and Burian, K., Investigations towards an artificial cochlea, Int. J. Artif. Organs, 2 (1979) 255-261.

- Hochmair-Desoyer, I. J., Hochmair, E. S., Burian, K. and Stiglbrunner, H. K., Percepts from the Vienna cochlear prosthesis, Ann. N. Y. Acad. Sci., 405 (1983) 295-306.
- House, W. F. and Edgerton, B. J., A multiple-electrode cochlear implant, Ann. Otol. Rhinol. Laryngol., 91, Suppl. 91 (1982) 104-116.
- House, W. F. and Urban, J., Long term results of electrode implantation and electronic stimulation of the cochlea in man, Ann. Otol. Rhinol. Laryngol., 82 (1973) 504-517.
- Kelly, J. L., Jr., Lochbaum, C. and Vyssotsky, V. A., A block diagram compiler, Bell Syst. Tech. J., 40 (1961) 669-676.
- Kiang, N. Y. S., Eddington, D. K. and Delgutte, B., Fundamental considerations in designing auditory implants, Acta Oto-laryngol., 87 (1979) 204-218.
- Kiang, N. Y. S. and Moxon, E. C., Physiological considerations in artificial stimulation of the inner ear, Ann. Otol. Rhinol. Laryngol., 81 (1972) 714-730.
- Klinke, R. and Hartmann, R., Auditory prosthesis: Basic physiology, Proc. XIIth ORL World Congr., Budapest, Hungary, 1981, pp. 509-513.
- Ling, D. and Nienhuys, T. G., The deaf child: Habilitation with and without a cochlear implant, Ann. Otol., 92 (1983) 593-598.
- Loeb, G. E., White, M. W. and Jenkins, W. M., Biophysical considerations in electrical stimulation of the auditory nervous system, Ann. N. Y. Acad. Sci., 405 (1983) 123-136.
- Markel, J. D. and Gray, A. H., Jr., Linear Prediction of Speech, Springer-Verlag, New York, 1976.
- Merzenich, M. M., Coding of sound in a cochlear prosthesis: Some

- theoretical and practical considerations, Ann. N. Y. Acad. Sci., 405 (1983) 502-508.
- Merzenich, M. M., Michelson, R. P., Pettit, R. C., Schindler, R. A. and Reid, M., Neural encoding of sound sensation evoked by electrical stimulation of the acoustic nerve, Ann. Otol. Rhinol. Laryngol., 82 (1973) 486-503.
- Merzenich, M. M. and White, M., Cochlear implants: The interface problem. In F. T. Hambrecht and J. B. Reswick (Eds.), Functional Electrical Stimulation: Applications in Neural Prostheses, New York, Marcel Dekker (1977) pp. 321-340.
- Mladejovsky, M. G., Eddington, D. K., Dobbelle, W. H. and Brackman, D. E., Artificial hearing for the deaf by cochlear stimulation: Pitch modulation and some parametric thresholds, Trans. Am. Soc. Artif. Int. Organs, 21 (1975) 1-6.
- Muller, C. G., Psychoacoustics in auditory implant research. In Proc. XIIth ORL World Congr., Budapest, Hungary, 1981, pp.531-535.
- Muller, C. G., Comparison of percepts found with cochlear implant devices, Ann. N. Y. Acad. Sci., 405 (1983) 412-420.
- Paliwal, K. K., Comparative performance evaluation of different pitch estimation algorithms for noisy speech, Acoustics Lett., 6 (1983) 164-166.
- Parkins, C. W., Cochlear implant: A sensory prosthesis frontier, IEEE EMB Magazine, June (1983) 18-27.
- Pfingst, B. E., Operating ranges and intensity psychophysics for cochlear implants. Implications for speech processing strategies, Arch. Otolaryngol., 110 (1984) 140-144.
- Pfingst, B. E., Frequency difference limens for electrical stimulation of the deafened inner ear. In ARO Abstracts, 8th Midwinter Research

- Conference, p. 107, 1985.
- Pfingst, B. E., Burnett, P. A. and Sutton, D., Intensity discrimination with cochlear implants, J. Acoust. Soc. Am., 73 (1983) 1283-1292.
- Pfingst, B. E., Glass, I., Spelman, F. A. and Sutton, D., Psychophysical studies of cochlear implants in monkeys: Clinical implications. In R. A. Schindler and M. M. Merzenich (Eds.), Cochlear Implants, Raven Press, New York, 1985, pp. 305-322.
- Ross, M. J., Shaffer, H. L., Cohen, A., Frenberg, R. and Manley, H. J., Average magnitude difference function pitch extractor, IEEE Trans. on Acoust., Speech and Signal Processing, ASSP-22 (1974) 353-362.
- Sachs, M. B. and Young, E. D., Effects of nonlinearities on speech encoding in the auditory nerve, J. Acoust. Soc. Am., 68 (1980) 858-875.
- Shamme, S. A., The representation and processing of speech in the auditory system. In ARO Abstracts, 8th Midwinter Research Conference, p. 78, 1985.
- Shannon, R. V., Multichannel electrical stimulation of the auditory nerve in man. I. Basic psychophysics, Hearing Res., 11 (1983a) 157-189.
- Shannon, R. V., Multichannel electrical stimulation of the auditory nerve in man. II. Channel interaction, Hearing Res., 12 (1983b) 1-16.
- Shepherd, R. K., Clark, G. M., Black, R. C. and Patrick, J. F., The histopathological effects of chronic electrical stimulation of the cat cochlea, J. Laryngol. Otol., 97 (1983) 333-341.
- Simmons, F. B., Electrical stimulation of the auditory nerve in man, Arch. Otolaryngol., 84 (1966) 2-54.
- Simmons, F. B., Mathews, R. G., Walker, M. G. and White, R. L., A functioning multichannel auditory nerve stimulator, Acta Oto-laryngol., 87 (1979a) 170-175.

- Simmons, F. B., Walker, M. G., Mathews, R. G. and White, R. L., Percepts and discrimination by auditory nerve stimulation: A summary of results and some proposals for nerve viability evaluation. In D. L. McPherson and M. S. Davis (Eds.), Advances in Prosthetic Devices for the Deaf: A Technical Workshop, National Technical Institute for the Deaf, Rochester Institute of Technology, Rochester, NY, 1979b, pp. 271-274.
- Sinex, D. G. and Geisler, C. D., Responses of auditory-nerve fibers to consonant-vowel syllables, J. Acoust. Soc. Am., 73 (1983) 602-615.
- Spelman, F. A., The cochlear prosthesis: A review of the design and evaluation of electrode implants for the profoundly deaf, CRC Crit. Rev. Bioengineering, 8 (1982) 223-252.
- Sung, W. Y. and Un, C. K., A high-speed pitch extractor based on peak detection and AMDF, J. Korea. Inst. Electr. Eng., 17 (1980) 38-44.
- Tong, Y. C., Black, R. C., Clark, G. M., Forster, I. C., Millar, J. B., O'Loughlin, B. J. and Patrick, J. F., A preliminary report on a multiple-channel cochlear implant operation, J. Laryngol. Otol., 93 (1979) 679-695.
- Tong, Y. C. and Clark, G. M., Absolute identification of electric pulse rates and electrode positions by cochlear implant patients, J. Acoust. Soc. Am., 77 (1985) 1881-1888.
- Tong, Y. C., Clark, G. M., Blamey, P. J., Busby, P. A. and Dowell, R. C., Psychophysical studies for two multiple-channel cochlear implant patients, J. Acoust. Soc. Am., 71 (1982) 153-160.
- Tong, Y. C., Clark, G. M., Seligman, P. M. and Patrick, J. F., Speech processing for a multiple-electrode cochlear implant hearing prosthesis, J. Acoust. Soc. Am., 63 (1980) 1897.
- Tong, Y. C., Dowell, R. C., Blamey, P. J. and Clark, G. M., Two-component hearing sensations produced by two-electrode stimulation in the cochlea

of a deaf patient, Science, 219 (1983) 993-994.

Un, C. K. and Yang, S.-C., A pitch extraction algorithm based on LPC inverse filtering and AMDF, IEEE Trans. on Acoust., speech and Signal Processing, ASSP-25 (1977) 565-572.

Van den Honert, C. and Stypulkowski, P., Spatial mapping of electrical excitation patterns within the cochlea derived from single fiber responses. In ARO Abstracts, 8th Midwinter Research Conference, p. 106, 1985.

Walsh, S. M. and Leake-Jones, P. A., Chronic electrical stimulation of auditory nerve in cat: Physiological and histological results, Hearing Res., 7 (1982) 281-304.

White, M. W., Formant frequency discrimination and recognition in subjects implanted with intracochlear stimulating electrodes, Ann. N. Y. Acad. Sci., 405 (1983) 348-359.

Appendix 1

RTI Patient Interface Description

RTI PATIENT INTERFACE DESCRIPTION

The Patient Interface is essentially a redesign of the existing UCSF Interface described in Mark White's thesis, Chapter 5. A redesign was initiated to take advantage of newer technology and to provide expanded system flexibility. The basic UCSF design features for patient safety precautions have been retained in the RTI design. The most significant design features include:

- a total of eight patient channels, each consisting of a computer-controlled stimulator for a bipolar electrode pair;
- independent channel functions with electrically floating grounds and isolated supplies;
- synchronization of all stimulus channel magnitude transitions with 50 usec. temporal resolution;
- integrated DAC's in the circuitry of each patient channel;
- optical isolation using a linear optically-driven FET thus reducing circuit complexity;
- continual monitoring of electrode voltages across any program-selected channel with an onboard ADC;
- electrode impedance measurement capability under program control between any two patient electrodes;
- patient connection or disconnection to or from the interface under program control.

PATIENT SAFETY DESIGN FEATURES

Patient safety design features parallel the UCSF design and are listed here for review:

- optical isolation of analog circuitry;
- dual output blocking capacitors of low capacitance and low leakage;

- patient disconnect relays on each channel operating under patient control or with an automatic timeout period of 50 usec.

The present RTI design does not incorporate the UCSF feature of a variable voltage battery supply. Instead, the high voltage supply for each patient channel consists of a standard modular supply, driven with an isolation transformer. The entire unit is further isolated with a medical-grade isolation transformer on the primary power support. This choice was made for the long term convenience of reduced maintenance and to avoid continued operating costs for battery replacement.

DG INTERFACE HARDWARE REQUIREMENTS

The Patient Interface communicates with the DG Eclipse system using the I/O capabilities of the DG Digital Control Unit, Model 50 (DCU/50). The DCU/200 may also be used. The DCU/50 is a user-programmable processor with 1024 words of onboard RAM and is capable of sharing 31,744 words of memory with the Eclipse via data channel communications. The DCU/50 essentially handles fast I/O between the Eclipse data bus and the DCU/50's own data bus. The DCU/50 communicates with the Patient Interface via a 16 bit parallel I/O interface DG 4066, which resides on the DCU/50 data bus. Patient channel data rates are sufficient to provide a channel frequency response of 20 kHz. DG hardware requirements for Patient Interface control include:

- one DCU/50 or DCU/200 I/O processor;
- one 4066 parallel I/O interface;
- one 4251 communications chassis for DCU data bus.

PATIENT INTERFACE SOFTWARE CONTROL

In general, the programmer has the capability to:

- (1) connect or disconnect the patient to or from the stimulation system;
- (2) on a channel-by-channel basis selectively change the stimulus magnitude every 50 usec. if desired, otherwise the previous magnitude value is maintained; for two channels only output may occur every 17 usec.
- (3) continually monitor the voltage across the electrode pair of a program-selected patient channel;
- (4) monitor impedance between any two program-selected patient electrodes;
- (5) perform automatic functional testing and calibration verification;
- (6) enable or disable the programmable clock onboard the interface, which times the DAC output conversions for all patient channels;
- (7) reset the Synchronization Error Flag after an error has been signalled from the interface;
- (8) determine identification number of the currently installed Electrode Selection Jumper Plug.

Operation of the Patient Interface with regard to each of these capabilities is detailed in the following paragraphs.

Program control of the Patient Interface is accomplished by passing 16-bit words via the DG 4066 parallel I/O interface. Each word consists of a 4-bit (most significant nibble) command field and a 12-bit data/function field. The 4-bit commands are summarized below.

CONTROL WORD (hex)	FUNCTIONAL DESCRIPTION
0***	Latch current data value (***) to patient chan 0
1***	" " " " " " " " 1
2***	" " " " " " " " 2
3***	" " " " " " " " 3
4***	" " " " " " " " 4
5***	" " " " " " " " 5
6***	" " " " " " " " 6
7***	" " " " " " " " 7
FSYZ	Logic Function Command

where

*** is current 12-bit data word (one's complement)

S is a subfunction code -

if S = 0, configure relays

Y = destination latch address

Z = latch value for relays.

if S = 1, enable/disable

if S = 2, enable/disable interface clock

if S = 3, ADC channel ON/OFF

if S = 4, ERROR indicator control ON/OFF

if S = 5, STIM ON indicator control ON/OFF

if S = 6, PATIENT RESPONSE PANEL LEDS

Data in bits 6-15

if S = 7; ADC0 READ
if S = 8; ADC1 READ
if S = 9; unused
if S = A; unused
if S = B; unused
if S = C, Reset Synchronization Error
if S = D, DISCONNECT Patient
if S = E, CONNECT Patient
if S = F, END OF COMMAND STRING identifier .

Patient Connection -

Control Word: FFXX

Function: Sets flag PC, which enables the capability of End Of Command (EOC) instructions to reset timing cycle of patient disconnect relay driver. Patient connection continues only as long as command strings terminated by EOC are being actively sent to the interface every cycle of the programmable clock (50 usec. typically).

CAUTION: All patient output channels should be zeroed prior to execution of this instruction.

Patient Disconnection -

Control Word: FDXX

Function: Set flag PD, which forces an immediate interface reset thus disconnecting the patient from the stimulation system. An automatic patient disconnect occurs if the timeout disconnect period timer has not been reset within the current clock, interval by the EOC command.

Channel Magnitude Control -

Control Words: #***, #***, #***,, #***, FFXX

where # is patient channel number (0-7)

*** is 12-bit one's-complement data value for DAC

FFXX is the End Of Command (EOC) control instruction.

Function: This instruction string transfers stimulus output magnitude values to each patient channel specified in the control word. These stimulus magnitude values are stored in data latches for each channel as the control word is received. The latched magnitudes are then transferred to the channel DAC's at the beginning of each programmable clock period, thus providing synchronization across all channels. If a stimulus magnitude value is not sent to a particular channel, then the previous latch value is retained for transfer to the DAC again. Consequently, presentation of a stimulus series requires only coding of changes of stimulus magnitude. Sustained values across one or more clock periods are automatically retained.

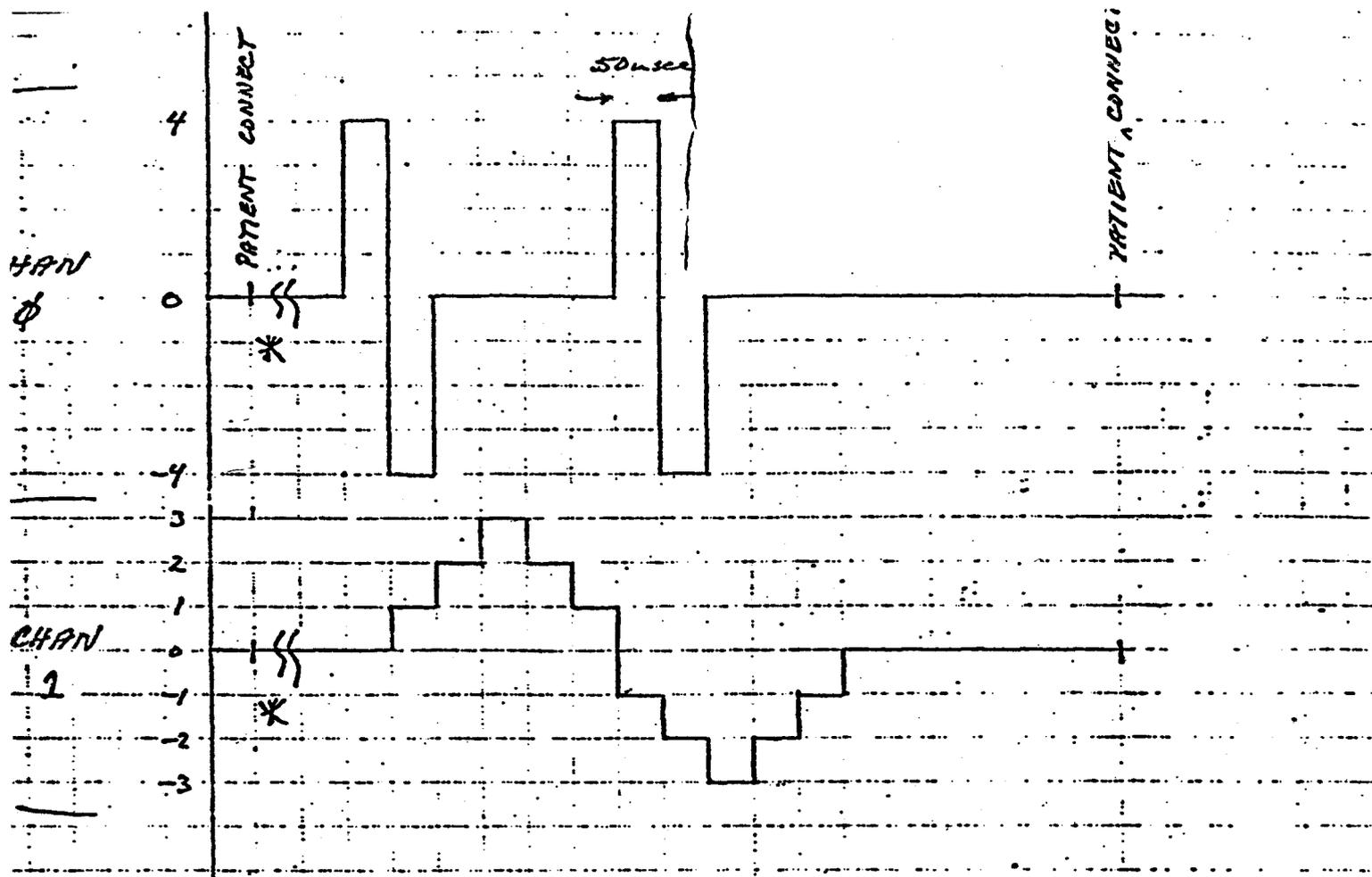
The EOC command must terminate the transmission of stimulus magnitudes for every clock period. The EOC command performs several functions:

- (1) the EOC allows a synchronization check between the Eclipse data transmission and the interface clock to ensure that all patient channel stimulus level updates have been received prior to the beginning of the next clock period. In the event that an EOC has not been received in time, the patient is automatically disconnected and an interrupt with the synchronization error flag set is issued to the Eclipse;
- (2) the EOC occurrence every clock period serves as an indicator of a flow of stimulus commands from the DCU, thus keeping the patient connected to the system. Should the data flow be interrupted, as

indicated by an EOC not being received within a clock period, the patient is automatically disconnected. This latter state is flagged as a synchronization error, as in (1) above;

- (3) the EOC, by itself, indicates a stimulus magnitude coding period during which no patient channel stimulus levels are modified. Stimulus delays to allow for relay contact bounce, while still maintaining patient connection, are achieved by a continuous string of EOC's. *~ , delivered ^{one} ~~more~~ per clock period*

The following page illustrates the stimulus coding required to output two different stimulus sequences simultaneously on two different channels. The Channel 0 output is a 50 usec. biphasic spike train, whereas the Channel 1 output mimics a continuously variable analog signal.



The command string to output this stimulus sequence would be:

RT)	CH0, 0	CH1, 1	EOC
	CH1, 4	EOC	EOC
	EOC	CH0, 4	EOC
	PATIENT CONNECT COMMAND		CH1, -1
	EOC	EOC	EOC
	CH0, 4	CH0, -4	EOC
	EOC	CH1, -2	PATIENT DISCONNECT COMMAND
	CH0, -4	EOC	(END)
	CH1, 2	CH0, 0	
	EOC	CH1, -3	
	CH0, 0	EOC	
	CH1, 2	CH1, -2	
	EOC	EOC	
	CH1, 3	CH1, -1	
	EOC	EOC	
	CH1, 2	CH1, 0	
	EOC	EOC	

* // indicates an interval in which a string of repeated EOC's, one every 50msec are required for a time delay during which patient disconnect relay box occurs (approx 20msec)

cont ↑

A-1-10 cont ↑

Patient Channel Electrode Voltage Monitoring -

Control Word: 9GPN

where G is patient channel number (0-7) for ADC Ground,

P is electrode number (0-15) for ADC Positive input,

N " " " " " " Negative " .

Function: To monitor the electrode voltage across a particular patient channel, the ADC input terminal connections are configured across that channel only. ADC data conversions are initiated at the beginning of every 50 usec. period with the result of the previous period made available immediately upon entry to the interrupt service routine in the DCU. Voltage monitoring may continue for the duration of stimulation. This command may be inserted into any control sequence before the EOC. Due to contact bounce of switching relays, valid data are not available for about 20 usec.

CAUTION: Ambiguous results will be obtained unless the following conventions are followed:

If M = desired patient channel number for voltage monitoring,

then

G = M (0-7)

P = 2M and N = 2M+1 (0-15)

or

P = 2M+1 and N = 2M (0-15)

Impedance Monitoring -

Control Word: AXXI

where I = 0 to exit impedance mode,

I = 1 to enter impedance mode.

Function: The impedance mode is a unique mode of operation that allows the measurement of the impedance between any two patient electrodes. The electrodes need not be in the same patient channel. Functionally, the impedance mode allows the programmer to direct the current stimulation from stimulator channel 0 to the electrodes to which the positive and negative inputs of the ADC are connected.

Therefore, to measure the impedance between electrodes Y and Z (0-15), the command sequence is:

Patient Disconnect	FXXX
Delay for relay bounce	Programmed Delay
Configure ADC for:	90YZ
G = 0, P = Y and N = Z	
Note: ADC ground is connected to the floating ground of stimulator channel 0.	
Enter Impedance Mode	AXX1
Patient Connect	EXXX
Delay for relay bounce with multiple EOC's in order for patient to remain connected	80CB
Begin output of stimulus current (***) on channel 0	0***,EOC
Read voltage values during interrupt processing at beginning of each 50 usec. period	

Repeat output and measure cycle	O***,EOC
" " " " "	" "
Complete measurement	
Disconnect Patient	FXXX
Delay for relay bounce	Programmed Delay
Exit Impedance Mode	AXXO

Automated Functional Checkout Routines

Utilizing the various stimulation and monitoring features of the interface, numerous options exist for automated interface checkout and calibration with known resistances in place of the patient. These routines will be specified later.

PATIENT ELECTRODE CONNECTION AND SELECTION

The RTI Patient Interface design departs from the UCSF Interface with regard to the selection of patient electrode configurations. The four channel UCSF system employs a relay matrix which allows complete versatility of electrode configuration under program control. However, as additional channels are added, the switching matrix size rapidly expands to cumbersome limits. For example, to provide full flexibility for an eight channel system would require 256 switching relays, as compared to the four channel system with only 16 relays. Excessive costs and circuit packaging problems make the relay matrix approach unattractive for an eight channel system. Considering that present encoding designs of stimulator systems do not require the ability to change electrode configurations during stimulation, a different electrode selection scheme has been adopted for the RTI design.

Essentially, a "poor/practical man's " switching matrix is utilized. This consists of a female, panel-mounted, multi-pin connector which has on one side the outputs from the current stimulators and on the other side the lines to the patient electrode disconnect relays. Electrode configurations are easily selected by plugging in a mating male connector, whose pins have been appropriately jumpered for the required interconnections. Rapid changes of electrode configurations are achieved by simply changing pre-wired jumper plugs. Extra pins on the jumper plugs may be wired so that the computer may verify that the appropriate plug is installed for the present protocol. The jumper plugs make attractive modules when standard cable hoods are installed on the male plugs. Jumper plugs are identified by number.

The impedance testing feature of the interface has been constructed so that full flexibility of electrode interconnections is available for impedance measurements under program control, regardless of the installed

jumper configuration. This enables spot checking of impedances between any two electrodes during patient testing.

Should program control of interconnections be required at a later time, an externally-mounted switching circuit could be built and used instead of the prewired jumper plug.

Appendix 2

Model of Field Patterns Produced by Intracochlear Electrodes

Model of Field Patterns Produced by Intracochlear Electrodes

Introduction

It is clear that the success of advanced speech processor designs is largely dependent upon the success with which neural elements can be predictably and discretely stimulated. Knowledge of electrical stimulation phenomena has been largely sought along two avenues. One is the description of VIIIth nerve firing in electrically-stimulated animal cochleas. The other is psychophysical experimentation with implanted patients. Both approaches are empirical and will require extensive experimentation before a good appreciation of the mechanisms is achieved. With regard to the human experiments, it is unlikely that identification of an optimum electrode would ever be achieved given the relatively small number of implanted patients, the biological diversity due to differing neuron survival patterns, and the variety of experimental stimulation procedures used to drive the electrodes. This point is evident when one considers the broad range of electrode configurations used in laboratories around the world. Unfortunately, there does not appear to be a single body of knowledge available with which these various strategies may be objectively evaluated and compared, other than the current approach of comparing the overall success of each respective prosthesis system. Consequently, we have initiated an effort to construct a biologically-authentic computer model of the physical structures and biophysical mechanisms thought to be involved in transduction of electrical stimulation to neural cochlear outflow. Our plan is to describe as accurately as possible the biological and physical parameters involved (i.e., tissue and electrode impedances, electrode positions and orientations, and normal and pathological neural characteris-

tics) and then to study the relationships of these parameters as they impact upon the stimulation of and the dynamic performance of the surviving neural elements within the implanted cochlea. Ultimately, we hope to achieve the following:

1. an identification of the most significant and sensitive factors in the design of an implantable electrode array;
2. an identification of the behavior of the neural elements depending upon their spatial position relative to the stimulating electrodes;
3. an identification of the behavior of the neural elements depending upon the temporal and magnitude characteristics of the electrical stimulation waveforms;
4. an identification of the factors contributing to channel interactions with emphasis on finding ways to reduce and/or exploit such interactions;
5. an estimation of the spatial distribution and temporal firing characteristics of the population of neural elements stimulated by any given electrical stimulus delivered to selected electrodes in the array;
6. the application of the above knowledge toward the design of an electrode configuration that optimally meets the clinical objectives of the overall prosthesis design;
7. the application of the above knowledge toward optimizing the transfer of speech information from the speech processor to the central nervous system, by best simulating normal cochlear transduction.

The following subsections describe the overall experimental approach, initial results to date, plans for future experimentation, and thoughts on the ultimate applications of the model predictions.

Two-Dimensional Cross-Sectional Model

The model consists of an iterative two-dimensional finite element description of a cochlear cross section containing the three scalae, the spiral ganglion, and a bipolar electrode pair in the scala tympani. The electrode pair represents the current USCF bipolar electrodes compressed into two dimensions. Grid points in the model are 20 microns apart and the two-dimensional sheet is assumed to be 20 microns thick. Presently, no assumptions are made relative to tissue characteristics in the planes parallel to the cross section, other than that complete symmetry exists. Potential distributions are computed by first defining a heterogeneous resistive plane which describes the resistive characteristics of the various tissues seen in cross section. The electrodes, with their associated resistances, are described in this plane as well. Fixed potentials are defined for the electrode regions, and the perimeter of the cross-sectional grid is assumed to be a neutral ground with a potential level midway between the potentials of the electrodes. Calculations consist of treating each grid point in the plane as a resistive node, surrounded by four adjacent nodes. Current node equations are solved to compute a new grid-point potential. This process is iteratively executed until the total absolute potential change, summed across all grid points, is less than 5% of the interelectrode voltage. With the present grid size of 120 by 120 points, one iteration lasts about 35 seconds, using the hardware floating-point processor of our Data General Eclipse computer. Approximately 300 to 400 iterations are required to achieve the error criterion.

This model differs considerably from the lumped-element models which presently figure heavily in the cochlear prosthesis literature. The lumped-element models focus upon the space constants along each of the scalae and provide limited insight into the tradeoffs between close longitudinal spacing of electrodes (or electrode pairs) and channel interactions. These models are of no real use in the prediction of neural excitation due to electrical stimulation and fail to provide insight into the biophysical phenomena occurring in the immediate proximity of the stimulating electrodes themselves. Moreover, human experimentation with multichannel electrodes, placed according to the lumped-element model results for minimizing channel interactions, reveal considerable electrode channel interactions in some patients. These channel interactions depend upon both spatial and temporal parameters and may correlate with the survival patterns of spiral ganglion cell dendrites in each patient. These results suggest that a more useful modeling approach would be one that accounts for both the spatial distribution of currents in the cochlea as well as the biological characteristics of the neurons, which underlie the temporally-dependent behavior of the system. It is important to note that any model will be limited in its predictive accuracy, but may yet serve a useful purpose in providing insight into the explanation of observed phenomena and the design of new experiments and stimulus techniques.

The two-dimensional, finite-element model is most attractive in that it allows the calculation of complex field patterns which exist in heterogenous structures of varying tissue types. For present purposes, the cochlear tissues are assumed to be purely resistive and isotropic. Anisotropic characteristics, which are known to exist in bone and myelinated nerve bundles, may be added later as the model is improved. The initial step in

the model description is the representation of the electrode array. Figure 1a shows the representation of a USCF bipolar electrode pair with a 90 degree interelectrode angle. The electrodes are assumed to be mounted in an insulator medium with the entire electrode assembly located in a homogenous resistive plane. Field potential patterns have been calculated assuming a fixed potential difference between the electrodes. The field distribution is indicated by equipotential contours (+ or - 1%) placed at 10% increments of the fixed electrode potentials. A current density diagram may be obtained by drawing contours perpendicular to the equipotential contours.

The purpose of the finite-element model is to describe the electrical nature of the cochlear tissues in cross section around the electrode. Figure 1b illustrates our plan for the final two-dimensional model. The cochlear cross section shown is a cartoon. Cross sections used for final calculations will accurately represent sections through actual cochleas at different turns. The anatomically-accurate cross sections will be entered into the computer by using a data tablet to digitize photographs of histological cross sections.

Initial Results

For the purpose of the present discussion, we assume that the field shown in Figures 1a and 1b approximates the actual field for the given cross section. This assumption will be tested when the anatomically-accurate cross sections are entered and fully modeled. The fields of ultimate interest are those which lie in the region of the myelinated spiral ganglion cells. Models of the response of myelinated neurons to electrical stimulation predict that sites of spike initiation occur where the maximum voltage gradients exist between adjacent nodes of Ranvier. Disregarding polarity considerations, it is evident that for the neuron shown in Figure 1b, the

initiated spike would begin in the dendrite region where the voltage gradients are the steepest, as opposed to the more proximal axon which lies approximately tangent to the equipotential contours. To obtain further insight into the nature of the voltage gradient, Figure 1c shows an inset which represents the calculated voltage gradients along a straight line drawn along the approximate course of the neuron. The y-axis range of the inset is equal to the potential difference between the two electrodes. Points A, B, C and D indicate positions along the neuronal axis. The potential gradients along the neuron, as shown in the inset, are the greatest between points B and C, indicating the predicted region of spike initiation. Although the conclusions of the model are obvious here, field patterns obtained with anatomically-accurate cross sections are expected to be different. In addition, the question arises as to whether or not other electrode sizes, positions, and/or impedances will have significant impact on the field patterns.

As an illustration of our initial exploration of these latter issues, Figure 2b shows the field calculations of the same conditions of Figure 1 (repeated in Figure 2a), with the electrodes modeled as point sources instead of the previous button-shaped elements. These conditions, for Figure 2b, are those normally assumed for a bipolar electrode pair modeled as an electrical dipole. Comparing Figures 2a and 2b, it is clear that the field patterns are quite different and in particular that the potential gradients are steeper in the vicinity of the neuron when stimulation is with the button-shaped electrodes. This indicates that there are significant near-field patterns around the actual bipolar electrodes that alter the potential gradients around the target neurons. Therefore, the assumption that the bipolar electrodes behave as an electrical dipole appears to be unfounded.

Monopolar stimulation of the cochlea may be also modeled as shown in Figures 3b and 3c. Figure 3b shows the field pattern when the upper-most electrode (black) is driven relative to a remote ground return electrode. The remaining electrode in the scala tympani is allowed to float. Figure 3c shows the field patterns when the lower-most electrode is driven in a similar monopolar fashion. Figure 3a repeats the previously-discussed bipolar field patterns for reference. It is evident from these fields that monopolar stimulation is far reaching in effect, resulting in greatly reduced specificity of stimulation. Comparing the results shown in Figures 3b and 3c indicates that significantly different stimulation results may arise depending upon which of the electrodes of the bipolar pair is driven in a monopolar fashion. For the same monopolar stimulus levels, the potential gradients along the neuron are much greater when the upper-most electrode is driven, Figure 3b. Significant current spread is a widely accepted notion in the cochlear prosthesis literature, yet the practice of monopolar stimulation is still widely used. With the model of monopolar stimulation, we hope to better evaluate the mechanisms active in other prosthesis designs.

As further demonstration of the ability to manipulate experimental parameters, Figures 4a, 4b and 4c show the field changes due to increasing the interelectrode angle from 90 degrees to 180 degrees for the bipolar pair. Note that the potential gradients along the neuron diminish as the angle increases. The 180 degree configuration approximates the Hochmair electrode used by the Austrian team. Figures 5a and 5b show the changes due to rotating the orientation of the UCSF electrode 45 degrees toward the spiral ganglion. The gradient diminishes slightly, but its position shifts toward the ganglion. This may be significant in cases of poor neuron survival. Figures 6a, 6b, 6c and 6d indicate the effects produced by

changing the position of the bipolar electrodes within scala tympani. Based on these results, electrode position within the scala is a significant design parameter (i.e., there are large differences in the field gradients, particularly for the conditions shown in panels 6c and 6d). Finally, Figures 7a and 7b depict the alterations in field patterns produced by reducing the bipolar electrode sizes and placing them closer together. The resultant field patterns are sharper and more circumscribed. Further discussion of the observed effects of these parametric changes is not warranted at present in that these calculations are only initial trial computations.

Future Modeling and Experimentation

The calculations of field distributions only provide a measure of the relative potential levels along the course of a spiral ganglion cell and its processes. Further calculations of action potential dynamics will be made by feeding the respective voltages of each node of Ranvier into a lumped-element model of a myelinated axon. McNeal's (1976) axon model, which consists of resistively linked Frankenhauser-Huxley nodes, will be used as a basic model. Two modifications will eventually be incorporated. One is the inclusion of lossy cable properties linking the F-H nodes instead of the purely resistive node interconnections of the McNeal model. This will enable accurate calculation of propagating action potentials. The other modification will be the description of the extra cellular node voltages as current sources in series with the extra cellular resistance. This will allow the node voltages to vary during calculated spike propagation. As a short note on the potential validity of the use of a mathematical neuronal model, it should be mentioned that this literature stems from the original Hodgkin-Huxley equations describing the giant squid axon. This original

work, and its many extensions, stand as one of the most remarkable successes in the modeling of a biological system. There is extensive work showing that these models accurately predict neuronal behavior in vivo.

A variety of possible electrode effects will be explored with the integrated field-neuron model, in addition to those already mentioned. In particular, the possibility arises that the relatively large button-shaped electrodes (see Figure 2) may exert substantial local (or near-field) effects on portions of a neuron, even though the two electrodes are being driven in a bipolar manner. Possibly, the effects of "anodal block" and/or "anodal break" may play a role during stimulation. These effects may effectively give rise to multiple generator sites as well as to blockage of cathodically-generated spikes. In addition, the temporal dynamics of the absolute refractory period, the relatively refractory period and accommodation phenomena further complicate the picture, but all can be computationally dealt with and evaluated in the model.

Validation of the model is of crucial importance to its ultimate utility. Clearly, one validation approach is to attempt to predict the results of numerous animal studies of VIIIth nerve responses to intracochlear stimulation. This will be clearly the most robust validation approach. Present expectations are that this validation will be straightforward. If the model proves effective here, then application of the model to the human cochlea is warranted. Michael Merzenich, during a recent visit to RTI, agreed to conduct animal and human experiments in his laboratories at UCSF to further test the model and/or its predictions, should circumstances warrant it.

Future Significance of the Modeling Approach

To summarize, the value of the two-dimensional model is largely focused on the following questions :

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Future Significance of the Modeling Approach

To summarize, the value of the two-dimensional model is largely focused on the following questions :

1. What are the relative effects of the heterogenous structure of the cochlea on the local field patterns in the vicinity of the spiral ganglion cells?
2. What are the optimal electrode configurations for intracochlear electrodes which discretely stimulate a limited population of cells?
3. What are the temporal stimulation characteristics of the neural elements in these electrode fields?
4. How do these characteristics limit speech-encoding strategies?

The greater issue of channel interactions is not directly addressed by the two-dimensional model. However, insight into the behavior of neurons local to a stimulating electrode pair will be essential toward understanding the factors controlling channel interactions. Finally, a three-dimensional model will yield the greatest insight into the channel interaction problem. The present two-dimensional iterative model can be expanded to three dimensions at great computational expense. Possibly, simplifying assumptions from the two-dimensional modeling, in particular that the cochlear tissue resistivities affect the actual field patterns only slightly, will reduce the simulation of the three-dimensional case to a relatively simple computation of linear summations at a point. If these simplifying assumptions are shown to be valid, only one three-dimensional computation of a single electrode pair would be required. Further discussion of the development of a three-dimensional model is deferred until complete results from the two-dimensional model are available.

Regarding the topic of channel interactions, several points may be made at this time.

(1) Judging from the presently limited two-dimensional model results, it is probable that the most significant channel interactions occur within the spiral ganglion itself. This hypothesis may be tested in at least two independent ways. One is to continue the modeling of potential fields to determine the loci of maximal summation of the field patterns of interacting channels. The experimental data useful in this approach are the "electrical" or spatial interactions obtained with synchronous channel stimulation. The other experimental approach is to carefully evaluate the temporal characteristics of channel interactions. A number of temporally-dependent mechanisms may underlie channel interactions. One is the strength-duration characteristic of a neuron. The details of how this factor affects channel interactions requires further thought, but it will undoubtedly be a significant factor in determining the optimum stimulation sequence for the encoding algorithm of the speech processor. A second temporal interaction between channels could arise from antidromically propagating spikes initiated by a basally located electrode pair (channel B), which collide with orthodromically propagating spikes initiated by a more apically located electrode pair (channel A). Possibly, a simple method of determining if these collisions occur would be to record the response measures (i.e. brainstem evoked responses and/or psychophysical reports; see also (2) below) to temporally delayed stimulations from channels A and B. If the responses to channel A followed by channel B are equal to the responses of channel B followed by channel A, there is little chance that significant spike collisions are occurring. It is difficult to say at this time how these possible effects may relate to the loudness-summation measures of channel interactions using comparisons of 0 degree and 180 degree phase conditions of two continuously stimulated channels (Shannon, 1984 preprint).

(2) Relative to the issue of measuring channel interactions, two methods are presently used in monitoring responses to stimulation in the human. These are the brainstem evoked response and the reporting of percepts by the patient. It appears that both of these measures may be biased when using the results to make inferences about the intracochlear mechanisms mediating channel interactions. Both techniques involve substantial processing of the activity of the VIIIth nerve before a quantifiable response is produced. As an alternative, we propose to simply record an "intracochlear" evoked response from a free pair of bipolar electrodes. This approach will provide a more direct measure of the specific gross activity of the spiral ganglion. This information may also be available at a higher signal-to-noise ratio, allowing reduced averaging and test time. Because these data may be less ambiguous and perhaps more may be collected in a given period, a more complete characterization of the channel interactions may be obtained. Interpretation these data in light of the modeling results could produce a physiologically-based model of how the prosthesis interfaces to the nervous system. Combining this knowledge with the information output from advanced speech processors, could, in turn, provide a basis for optimizing the stimulation strategy. Unique strategies, optimized for individual patients, may also be possible.

The RTI patient interface is being configured for this capability. In particular, electrical artifact rejection circuits are being included in the programmable ADC for monitoring channel activity.

References

McNeal, D. R., Analysis of a model of excitation of myelinated nerve,

IEEE Trans. Biomed. Eng., 23 (1976) 329-337.

Shannon, R. V., Loudness summation as a measure of channel interaction in a

cochlear prosthesis, 1984 preprint.

A-2-15

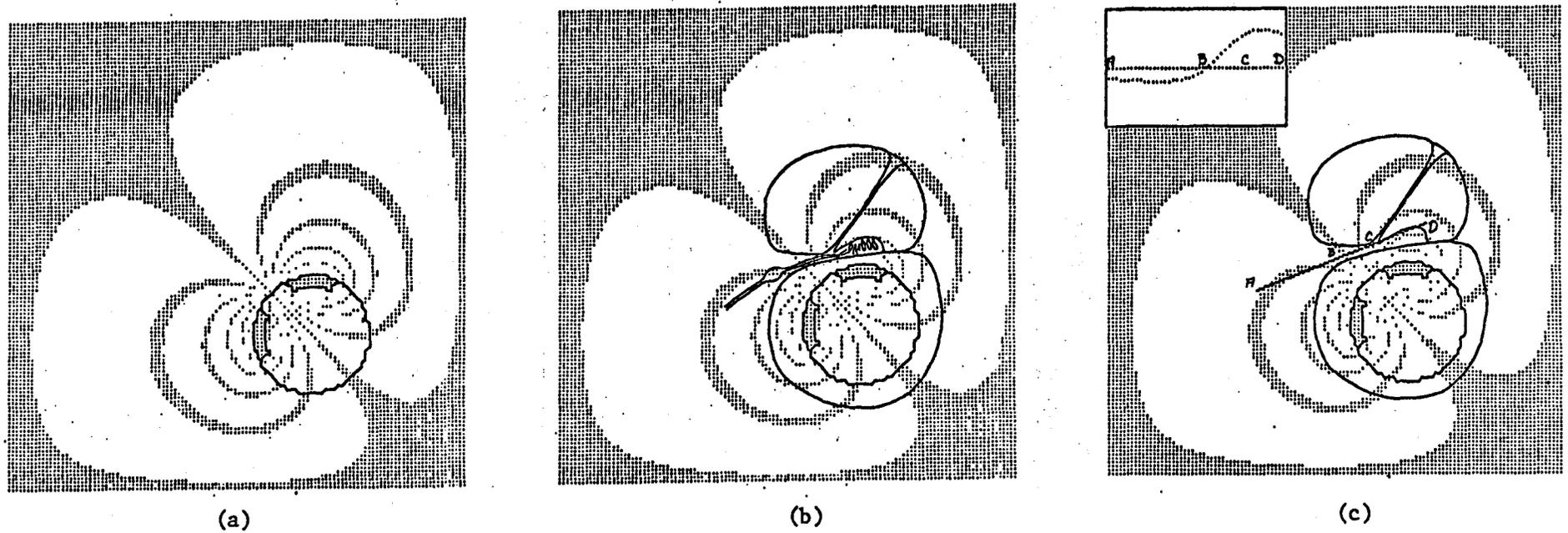


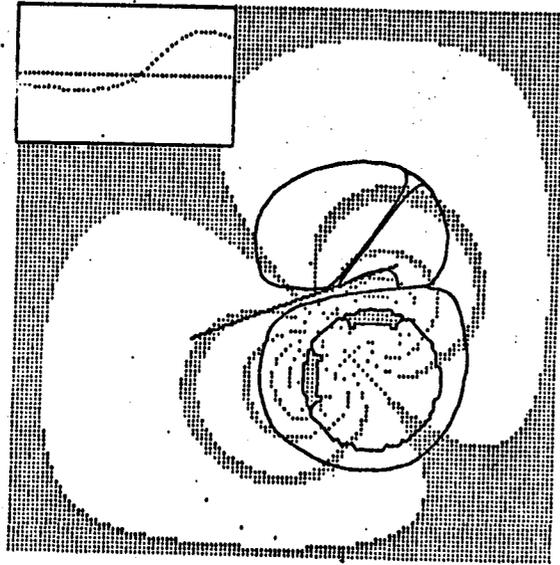
Figure 1. Description of two-dimensional finite-element model.

(a) UCSF bipolar electrode pair and computed field pattern

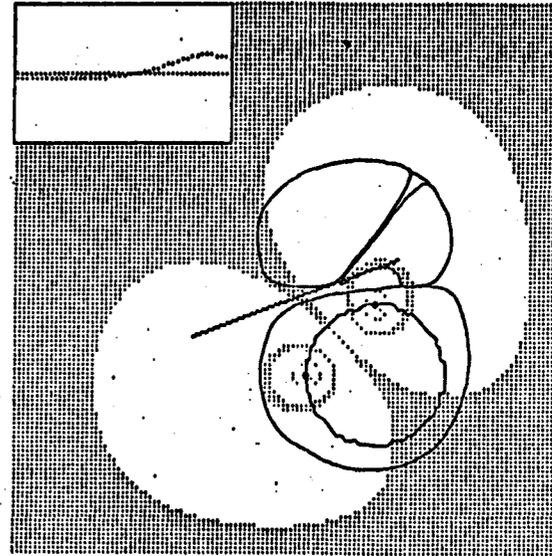
(b) Electrode and field pattern with overlay of cochlear cross section

(c) Same as (b) but with inset showing potential levels along neuron (see text)

A-2-16



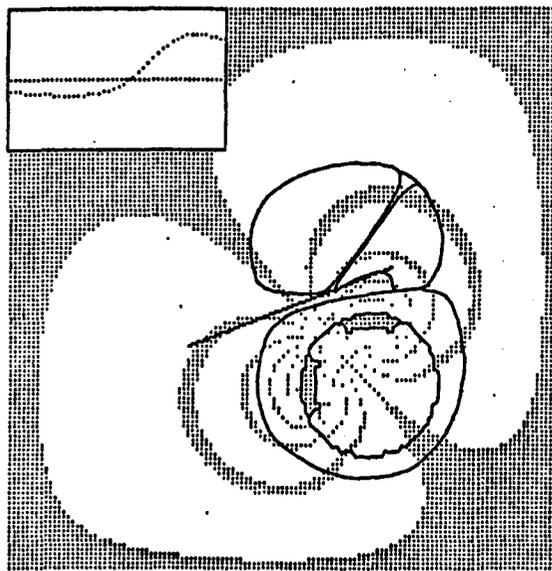
(a)



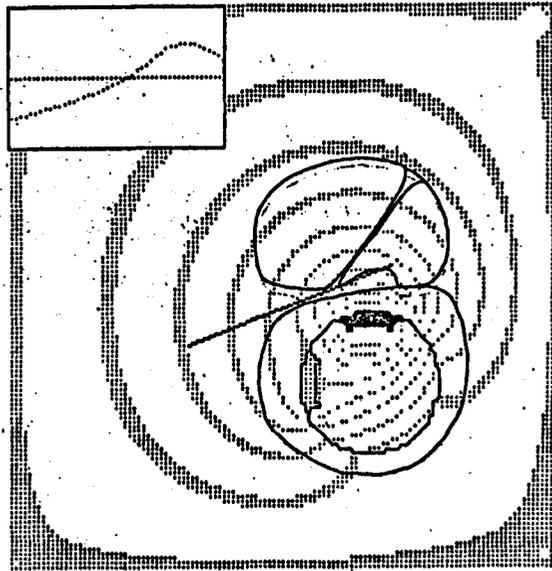
(b)

Figure 2. Field patterns for an actual bipolar pair (a) and a true dipole (b) configuration.

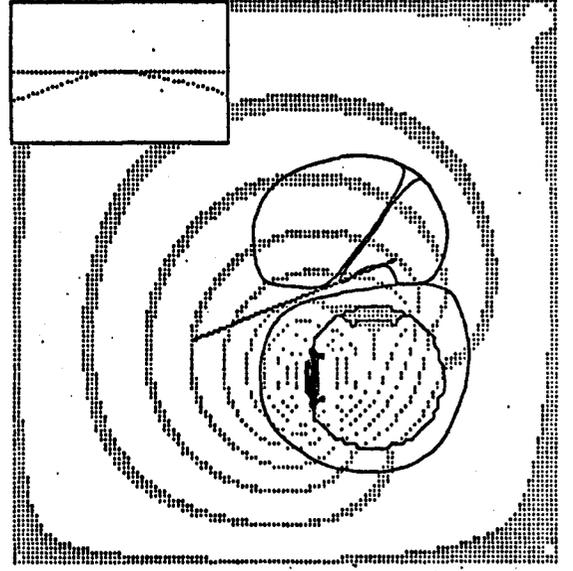
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(a)

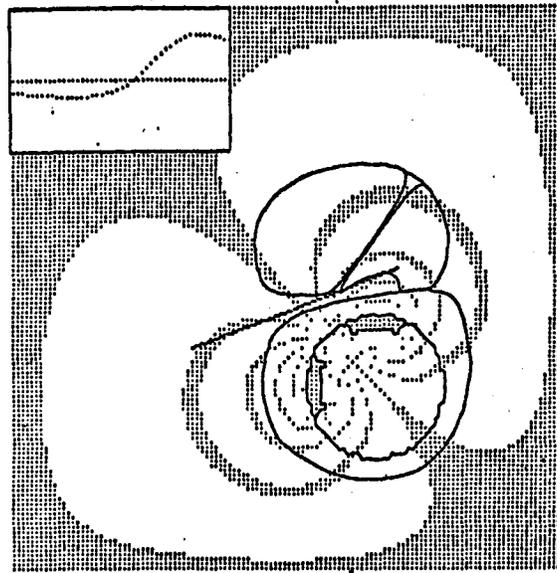


(b)

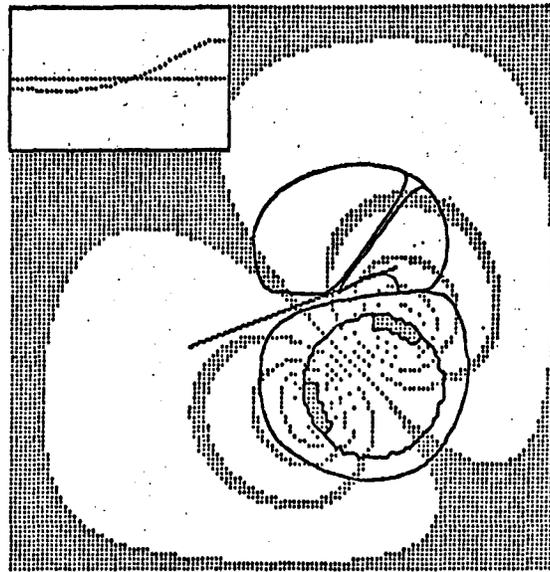


(c)

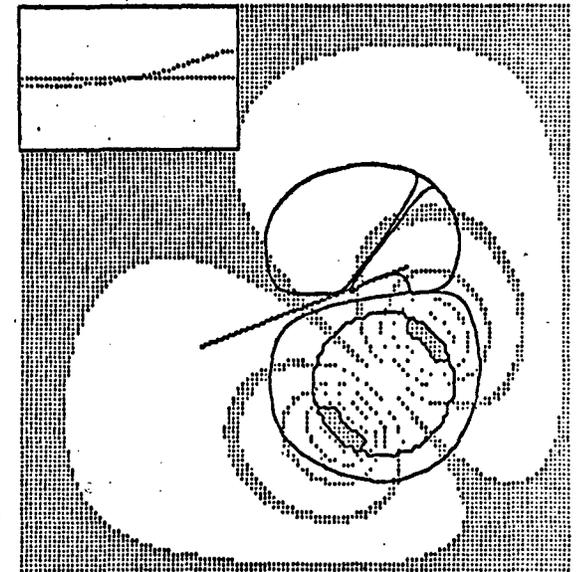
Figure 3. Field patterns for standard bipolar (a) and monopolar (b and c) configurations.



(a)



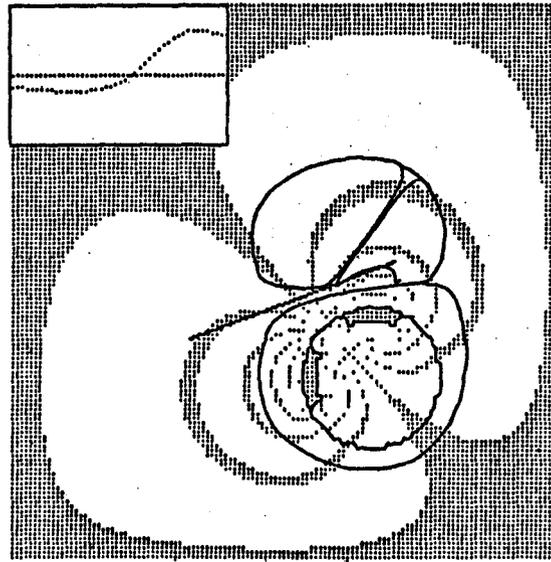
(b)



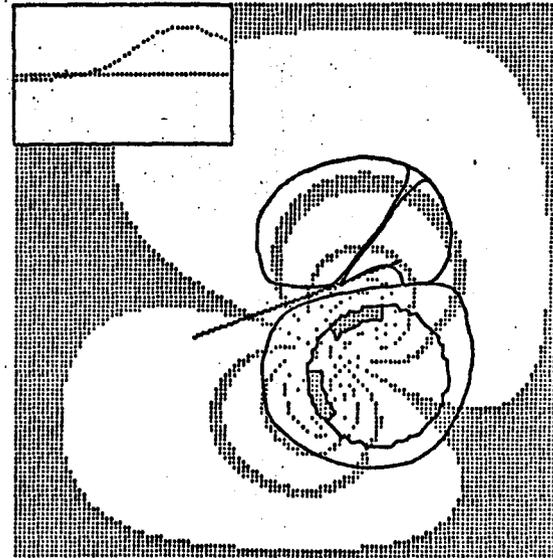
(c)

Figure 4. Field patterns for bipolar electrodes configured with differing interelectrode angles of 90° (a), 135° (b), and 180° (c).

A-2-14



(a)



(b)

Figure 5. Field patterns for standard bipolar electrodes configured with different rotational positions of 0° (a) and 45° (b) toward the spiral ganglion.

A-2-20

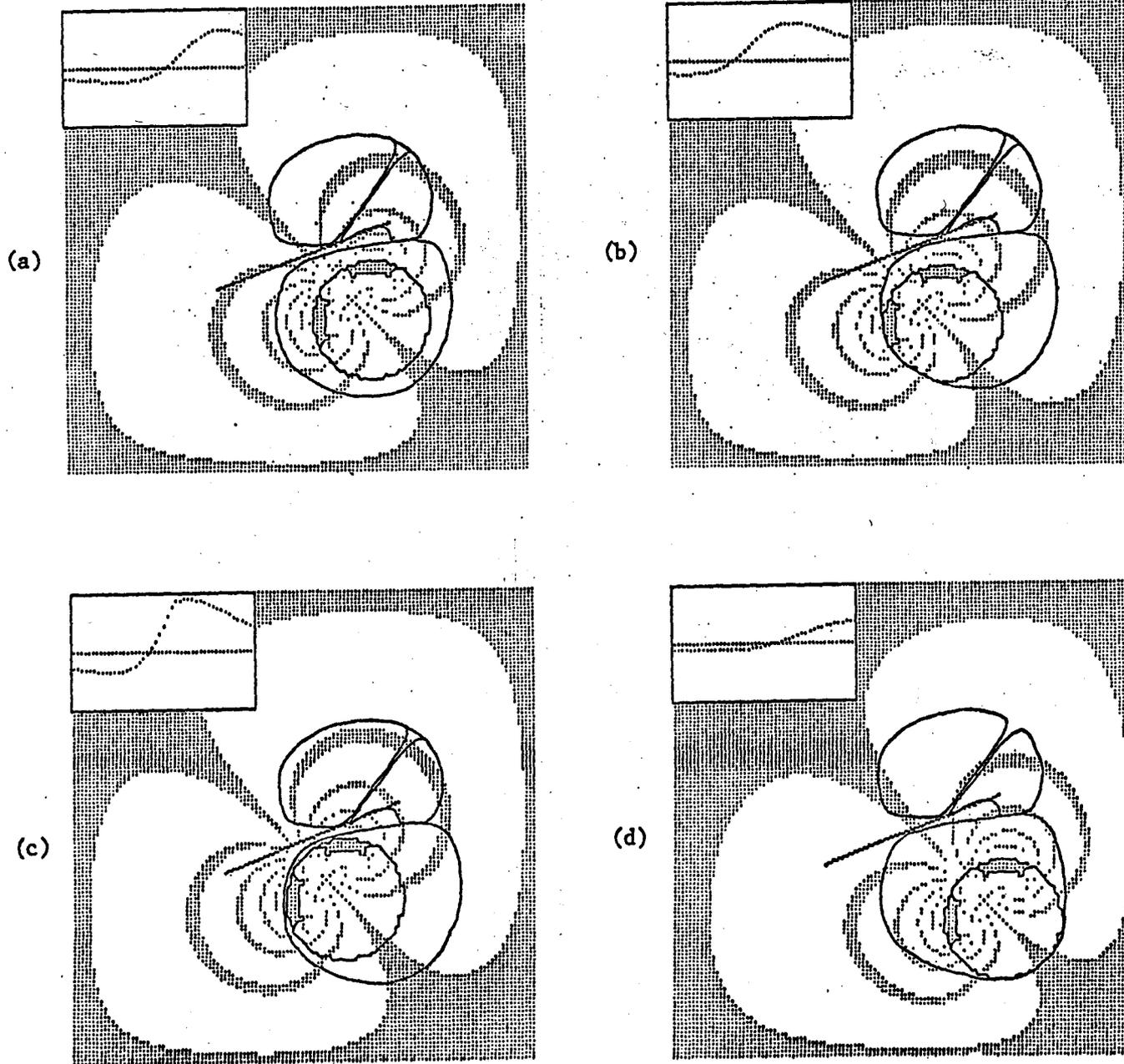


Figure 6. Field patterns for the standard bipolar pair located at different positions within scala tympani.

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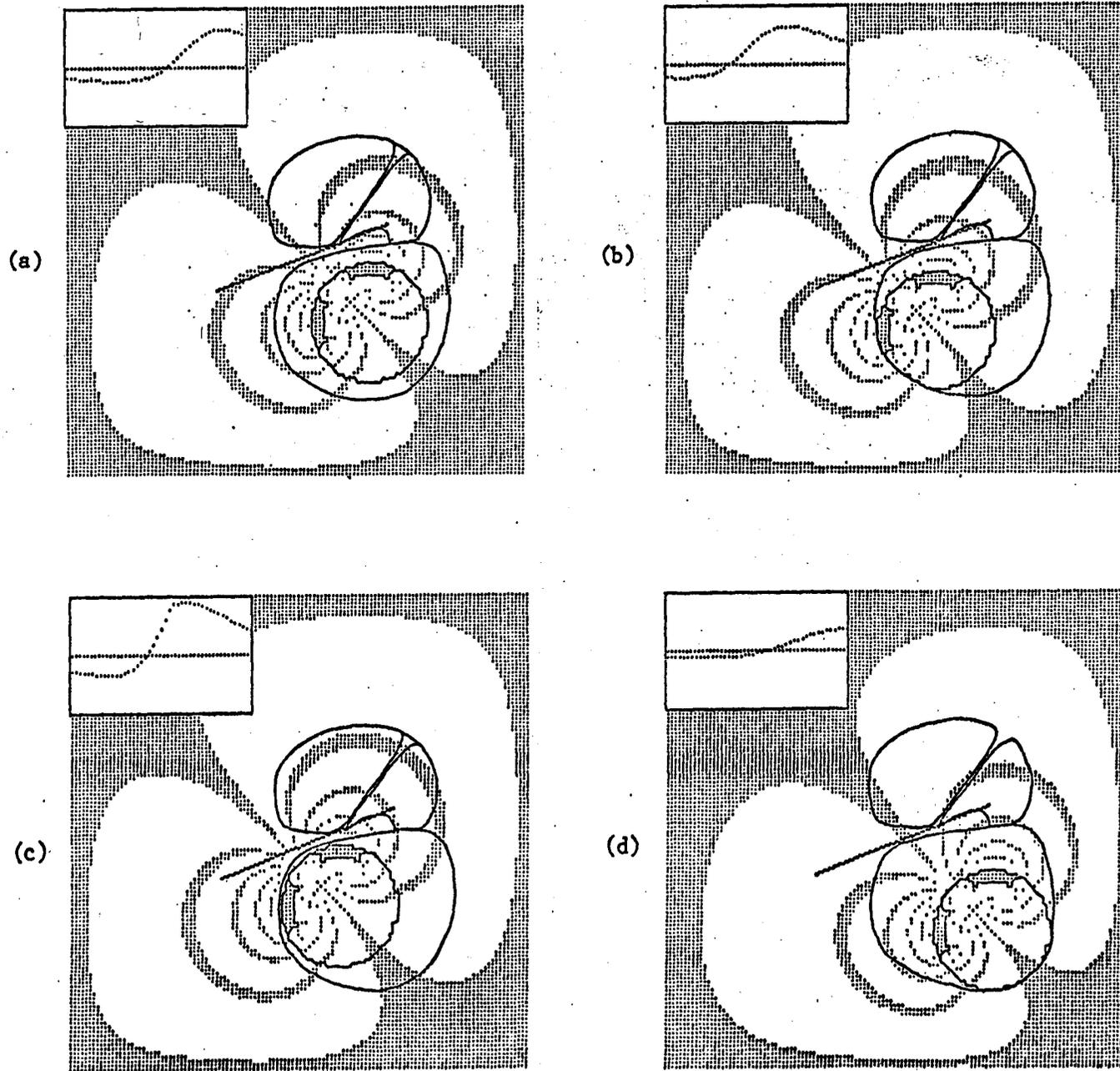
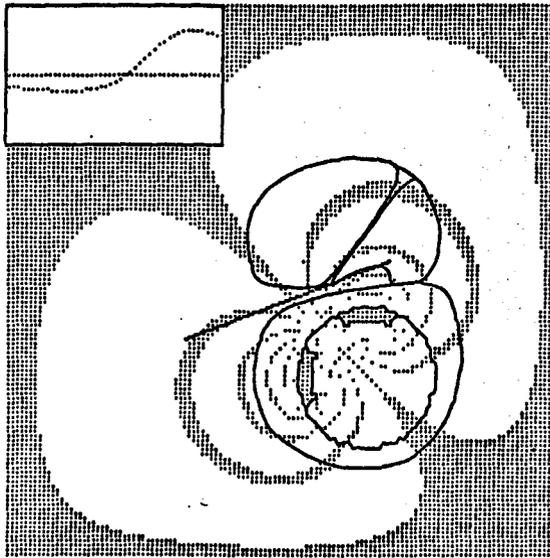
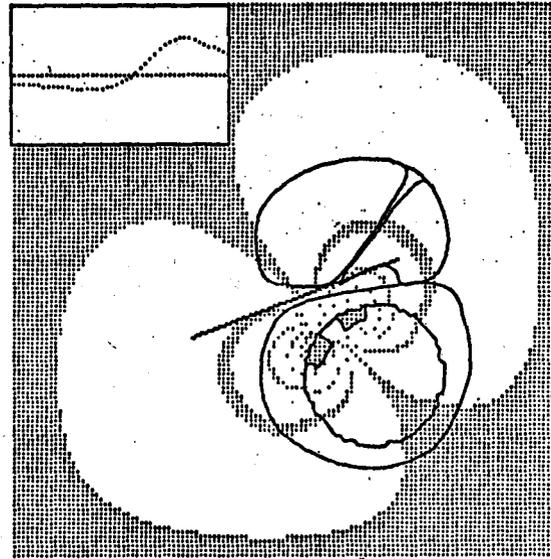


Figure 6. Field patterns for the standard bipolar pair located at different positions within scala tympani.

A-2-22



(a)



(b)

Figure 7. Field patterns for the standard bipolar pair (a) as compared to a smaller and more closely spaced bipolar pair (b).

Collaboration Between UCSF, Storz Instrument Company, RTI and DUMC

Mike Merzenich and two representatives from Storz Instrument Company, Dave Calvert and Steve Hutchison, visited RTI and Duke on March 21 to discuss Duke's potential participation as an "experimental collaborator" in UCSF's program to develop the next generation of multichannel auditory prostheses. Dr. Merzenich outlined the present status of the the four-channel UCSF prosthesis to the group at Duke and described the plans UCSF and Storz have for experimental and clinical collaborators. The suggestion was made that Duke participate as both, and that all four parties work closely together to conduct parallel tests at UCSF and Duke for evaluation of speech-processing strategies for multichannel prostheses. This suggestion was adopted by the group and preliminary arrangements have been made to start in earnest the new program at Duke. We expect that surgeons and audiologists from the Duke team will travel to San Francisco in the next month or two for training on the implant procedure and on evaluation and rehabilitation of patients. We hope that our first implant at Duke will be performed this Fall. A percutaneous connector will be used for this and subsequent patients. This connector will provide direct access to all electrodes in the implanted array so that we can duplicate at Duke the computer-based tests of speech-processing strategies we will be conducting at UCSF. If funds can be identified to support various aspects of the experimental tests at Duke, then the number of patients included in the present project could be approximately doubled. Additional support is required for the following: (1) installation and upgrading of a "spare" RTI Eclipse computer at Duke; (2) construction of an additional interface between the Eclipse and patient electrodes, for use at Duke; (3) computer supplies and maintenance associated with the conduct of tests at Duke; (4)

Appendix 3

Two Hypotheses of Multichannel Speech Processing Schemes

Based on Psychophysical Data from Implant Patients

from

RTI's "Best and Final Offer" submitted in response
to a request for additional information regarding
NIH RFP No. NIH-NINCDS-83-06, July 5, 1983.

REQUEST

1. "Supply further information on how speech cues derived from your speech processing will be translated to specific cochlear implant stimulus parameters and stimulus electrodes based on presently known psycho-acoustical results from multichannel cochlear implants."

RESPONSE

A. Introduction

Two designs of speech processors were presented in our original proposal to indicate ways in which parameters of speech essential to intelligibility could be encoded for electrical stimulation along an array of electrodes implanted in the scala tympani. These designs were intended to serve as examples of how the speech-encoding problem could be approached, and perhaps solved. In our response to the above request for further information, we will describe in greater detail how speech parameters are derived in the "front ends" of the two processors, how these parameters are translated into stimuli at the electrodes, and how selection of these parameters and stimuli is consistent with present knowledge of the psychophysical performance of implant patients. Before proceeding with this more detailed presentation, however, we would like to remind the review panel of the alternate approaches briefly mentioned in our original proposal on pages 5, 6, 22 and 35-38. These approaches, along with other processors proposed by other groups, are all worthy of careful evaluation. Therefore our guiding philosophy is not to champion one or a few approaches (however logical or well founded they may seem), but rather to consider a broad range of reasonable possibilities. This philosophy leads to a general plan with the following tasks: (1) identify and contrast the various approaches for solving the speech-encoding problem; (2) build a general-purpose, computer-based stimulator that is capable of rapid and practical emulation of all these approaches in software; and (3) evaluate all the approaches in tests with single subjects so that meaningful comparisons of performance can be made.* We believe all three tasks of this plan are important, if not essential, for the successful development of speech processors for auditory prostheses.

* Evaluation of all approaches in tests with single subjects is the only way to control fully for inter-subject differences in pathology (i.e., differences in the densities and loci of surviving neurons and possible differences in the integrity of central auditory structures), electrode type (e.g., there are many types of scala-tympani implants, including the scala-filling type used by the UCSF group; the "free-floating" types used by the Melbourne, Los Angeles, Utah and Vienna groups, and the multiple-fenestra types used by the Paris and Los Angeles groups), and apposition of individual monopolar or bipolar-pair electrodes to excitable tissue. These differences among subjects, along with differences in the testing procedures used, have made comparisons of speech processors developed in different laboratories tenuous at best.

B. Two Examples of Speech-Processor Design

Figure 1 shows the general scheme underlying the two speech processors presented in our original proposal. The speech analyzer is no different from that used in conventional vocoder systems. Its purpose is to extract from the input waveform a few parameters known to be essential in transmitting intelligible speech over data links of limited bandwidth. These parameters include the frequencies and amplitudes of the first 3 formants, the pitch of the voice in voiced speech sounds, and a binary indication of whether the present input is voiced or unvoiced (Flanagan, 1972).

While there is little disagreement on the set of extracted parameters essential for transmitting intelligible speech, there is disagreement on which parameters might be most essential and on how these parameters might be best translated into electrical stimuli for presentation along an array of electrodes implanted in the scala tympani or modiulus. The problem of selecting speech parameters and then encoding them into electrical stimuli is difficult, for several reasons. First, any scheme must be consistent with the narrow bandwidth of information that can be conveyed over the cochlear prosthesis. For a 15 channel prosthesis with a spatial resolution equal to that of the UCSF scala-tympani implant, Klinke and Hartmann (1981) have estimated a total channel capacity of between 1500 and 1800 bits/s. The information rate of all the extracted speech parameters mentioned above is around 1500 bits/s. Therefore, in a patient with good survival of neurons over all electrodes of the implant, a speech encoder may be able to map the parameters onto the limited perceptual space of the patient if the full range of available percepts is exploited. The translation of speech parameters into electrical stimuli must be done in a way that will produce distinct and separable percepts for each significant change in the speech parameters. Unless the percepts "track" the parameters with a minimum degree of accuracy (we will get into the numbers for this later), intelligibility will certainly suffer. Thus the encoding scheme has to be carefully tailored to the psychophysical performance of implant patients.

For patients whose psychophysical performance falls short of the ideal just indicated, hard choices must be made on which subset of speech parameters to encode. Inasmuch as most candidates for cochlear implants are likely to have at least some loss of neural tissue at sites where electrical stimuli can be delivered, and inasmuch as present cochlear prostheses have fewer than 15 channels, many investigators have made these choices in their designs. The encoder designed by the Melbourne group, for example, presents information to an array of nine electrodes on the frequency and amplitude of the second formant, the voice/unvoice distinction, and voice fundamental frequency during voiced speech sounds (Tong et al., 1980). Although encouraging results have been obtained with this approach, especially when patients use their prostheses in conjunction with lip-reading (Clark et al., 1981; Clark and Tong, 1982), it is not surprising that the greatest number of errors found in discrimination tests is between words with the same second formant and different first formants (Parkins, 1983). Thus, a price is paid for deletion of any of the parameters known to be essential in transmitting intelligible speech.

A-3-4

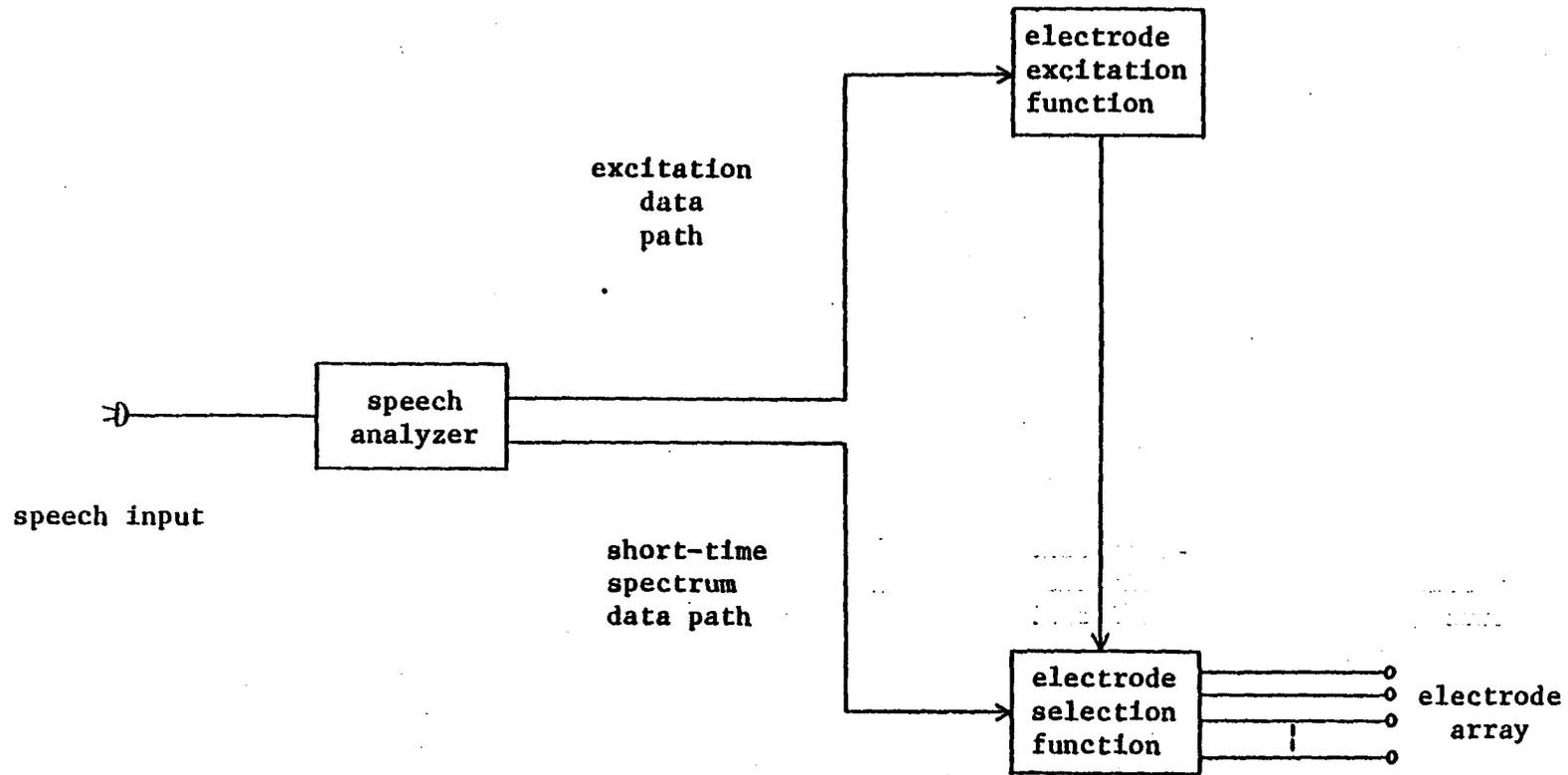


Figure 1. General scheme of the vocoder approach as applied to auditory prostheses.

Another major difficulty in the design of speech processors for cochlear prostheses is in devising a scheme that will not only map speech parameters onto the restricted perceptual space of implant patients but will present an input that the central auditory system can interpret. By this we mean that the encoding of speech parameters at the auditory nerve should be consistent with the decoding of these parameters in the central pathways. Take, for example, the problem of encoding formant frequencies. From single-unit studies, in which acoustic stimuli were used, we know that these frequencies are reflected in the discharge patterns of auditory nerve fibers in at least two ways: (1) the average rate of discharge as a function of distance along the cochlear partition follows the envelope of the acoustic spectrum for low-intensity speech sounds (Kiang and Moxon, 1972; Kiang et al., 1979) and (2) responses of neurons with best frequencies approximating the formant frequencies are synchronized to "preferred" phases of the time waveform over a broad range of intensities (Sachs and Young, 1979). The cochlear nucleus thus receives rate and periodicity information, which the central auditory system might use to infer formant frequencies. If a speech processor for a multichannel cochlear prosthesis somehow presented the formant frequencies in a code that mimicked only the rate profile found in normal hearing, then the success of this approach would depend on whether the patient could perceive changes in the rate profile and on whether the rate profile alone is adequate for decoding formant frequencies. If periodicity or periodicity and rate information is required by the central auditory system for reliable detection of formant frequencies, then the speech processor that encodes formant frequencies in terms of rate information only is likely to fail unless extensive training can "teach" the auditory system to use another input. Thus, an arbitrary mapping of speech parameters onto perceptual dimensions, while difficult enough in itself, is no guarantee that intelligible perception of speech will result.

With the above considerations in mind, we are now in a position to discuss in greater detail the two speech processors presented in our original proposal. As mentioned before, speech parameters are extracted using techniques established in the development of conventional vocoders. Translation of these parameters into electrical stimuli for the implanted array of electrodes follows the two paths leading from the output of the speech analyzer in Fig. 1. Parameters describing the formant structure of speech control the selection of electrodes to receive stimuli (labeled "short-time spectrum data path") and parameters describing the excitation of the vocal tract control the timing of biphasic pulses to be delivered to the selected electrodes (labeled "excitation data path"). The rationale for the explicit separation of electrode excitation and electrode selection functions is based on (1) the first-order independence of vocal-tract excitation and transmission in speech production (Flanagan, 1972) and (2) the partial independence of pitch percepts elicited with cochlear implants according to periodicity of stimulation and electrode position. Specifically, for a given electrode (either monopolar or a single bipolar pair) and intensity of stimulation, the percept of pitch follows the frequency of sinusoids, or the rate at which pulses are delivered, up to about 300 Hz (Bilger et al., 1977; Diller et al., 1980; House and Urban, 1973; Merzenich et al., 1973; Mladejovsky et al., 1975; Simmons, 1966; Tong et al., 1982). The difference limens (DLs) for this repetition or "volley" pitch are often 5% or less for frequencies below 200 Hz, but DLs rise rapidly as frequency is increased

much beyond 200 Hz. Also, results of scaling and matching experiments indicate that, while pitch corresponds to rate for stimulus frequencies up to about 200 Hz, it either accelerates to very high values (Eddington et al., 1978; Simmons, 1979b) or does not increase (Tong et al., 1979) thereafter. Useful encoding of speech parameters along the dimension of volley pitch percepts is therefore probably limited to frequencies below 200 to 300 Hz.

When the frequency and intensity of electrical stimuli are held constant, but the site at which stimuli are delivered is varied, distinct tonal sensations are evoked that can be ranked according to the electrode's position along the cochlear partition (Chouard and MacLeod, 1976; Eddington et al., 1978 and 1980; Hochmair et al., 1979; House and Edgerton, 1982; House and Urban, 1973; Mladejovsky et al., 1975; Tong et al., 1982 and 1983) or location within the auditory nerve (Simmons, 1966; Simmons et al., 1979a and b). To the extent that two mechanisms of pitch perception are involved in the complex sensations evoked by electrical stimuli (i.e., "place" and "volley" pitch), an efficient scheme for speech encoding might be to have the periodicity of electrical stimuli convey information about the excitation of the vocal tract, and to have the locations and number of electrodes at which these stimuli are delivered convey information about the configuration of the vocal tract. In this way a parsimonious description of speech sounds can be transmitted over an array of electrodes without exceeding the narrow dynamic range for intensity of electrical stimulation. The assumptions inherent in this general approach are that different electrodes in the implanted array evoke separate and independent percepts, that these percepts can be scaled in a tonotopic order according to place of stimulation, and that the information is presented in a form that can be decoded by the central auditory system.

Figure 2 shows a block diagram of one of our two designs that implements in hardware the general approach reviewed above. Two paths lead from the microphone input on the extreme left-hand side of the figure. The first components in the upper path are speech-analysis modules to extract parameters describing the excitation of the vocal tract. Specifically, the voice pitch extractor determines the fundamental frequency of the voice in voiced speech sounds, and the voice/unvoice discriminator separates unvoiced consonants from voiced consonants and vowels.

These extracted parameters describing vocal tract excitation are then used to control the times at which stimuli are delivered to all selected electrodes in the array. Stimulation at the fundamental frequency is provided when voiced speech is present at the microphone input, and randomly-timed stimuli are produced when unvoiced speech is present at the microphone input. Typical waveforms and the circuit configuration for the encoding of unvoiced speech are shown in the upper-right portion of the figure. The 1-kHz filter is inserted between the noise source and monostable multivibrator (labeled "schmitt/one shot") to prevent the occurrence of sequential pulses separated by less than 1 msec. This feature locks out the unnecessary delivery of stimuli during the absolute refractory period of previously-stimulated neurons.

The final signal delivered to the selected electrodes for both voiced and unvoiced sounds is a train of charge-balanced, biphasic pulses. Re-

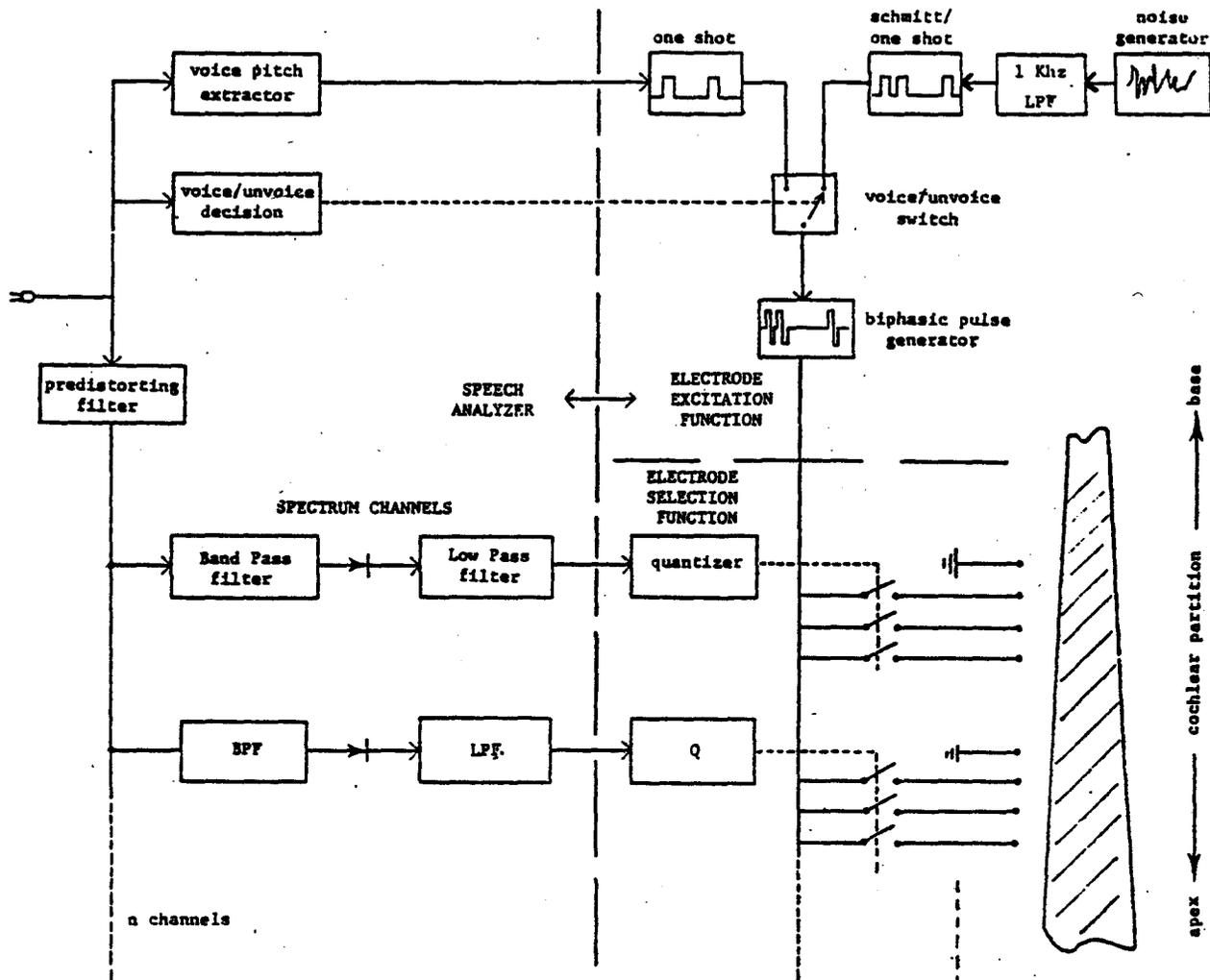


Figure 2. Channel vocoder approach as applied to auditory prostheses.

sults of several studies have demonstrated that use of such pulses minimizes pathological changes in tissue induced by chronic intracochlear stimulation (see, e.g., Shepherd et al., 1983; Walsh and Leake-Jones, 1982).

The lower path from the microphone in this spectrum-channel realization of a speech processor for a multielectrode prosthesis leads first to a "predistorting" filter and then to a bank of bandpass filters. The function of the predistorting filter is to emphasize the high-frequency part of the speech spectrum so that, over long averaging intervals, a nominally-flat spectrum is presented to the bank of bandpass filters. This operation of spectral tilting enhances the signal-to-noise ratio for detection of the second and third formants in vowels and for detection of consonants, which generally have low intensities relative to the vowels and have most of their energy in the high-frequency part of the speech spectrum.

As in conventional vocoders, the function of the bandpass filters is to extract the short-time spectrum of incoming speech. The energy in each band is sensed at the output of each filter with a rectifier and low-pass filter. A typical cutoff for the low-pass filter would be in the range of 20 to 30 Hz, reflecting the maximum rates at which speech articulators can move. The selection of break frequencies for the bandpass filters would depend, of course, on the number of filters and the total range to be spanned. Psychophysical studies of vocoder performance with normal-hearing subjects suggest that the bandwidths of the filters should be no more than 2/3 octave and the number of filters should be no fewer than 10 (Flanagan, 1972).

Translation of the information contained in the short-time spectrum into spatial patterns of electrical stimulation in the inner ear is performed by the quantizers in the lower-right portion of Fig. 2. The quantizers first sense the output energy in each band of frequencies from the bank of bandpass filters and then control the local extent of excitation according to a rule that maps the wide dynamic range of speech onto the narrow dynamic range of electrically-evoked auditory sensations. Loudness in each channel of bandpass frequencies will be least when the electrode closest to the "ground" or reference electrode is connected (by the quantizer) to the output of the biphasic pulse generator and greatest when the electrode farthest from the reference electrode is connected to the output of the biphasic pulse generator. The rule for mapping defines the compression characteristic for keeping the transformed outputs of the bandpass channels within the dynamic range of perception. This rule can take several forms, including non-instantaneous compression to mimic adaptation effects in the normal auditory periphery.

The arrangement of bipolar electrodes in the design shown in Fig. 2 was originally selected with two empirical observations in mind: (1) faithful transmission of formant frequencies is extremely important for the encoding of intelligible speech (Flanagan, 1972) and (2) Simmons has reported that, for a given electrode and repetition rate, increases in stimulus intensity are always accompanied by increases in perceived pitch for the modiolar prosthesis (Simmons, 1966; Simmons et al., 1979b). Apparently, the central auditory system weights the evoked activity of neurons with high best frequencies to infer pitch when many units are responding to

modiolar stimulation. The mechanism for this pitch perception is unknown. However, assuming the phenomenon of increased pitch with increased intensity also has significance for scala-tympani implants, our first thought was to position electrodes along the cochlear partition so that each bandpass/quantizer channel would have one reference electrode at the "place" corresponding to the center frequency of the bandpass filter. The intensity of each channel output would then be encoded by selecting the appropriate active electrode to recruit progressively more fibers in an apicalward direction for increases in perceived loudness. In this way the edge of stimulated neurons with the highest best frequencies would not move with changes in the spread of excitation. The idea is to prevent the spread of excitation in both the apicalward and basalward directions that occurs when intensity is encoded by stimulus amplitude alone. Thus, our objective is to maintain the perceived pitch of speech formants throughout the range of perceived formant loudnesses.

A curious consequence of this encoding scheme is that the spatial pattern of stimulation is opposite to the pattern found in normal hearing. That is, the amplitude of vibrations produced along the basilar membrane by acoustic sinusoids in normal hearing gradually increases from the base to a point of maximum vibration and then decreases precipitously thereafter. Thus, if we were to mimic the gross pattern of excitation found in normal hearing, the reference electrode would be the most apical in each set of electrodes rather than the most basal. To compound the dilemma of electrode arrangement, there is much evidence that the central auditory system attends to the sharp edge of apical falloff in basilar membrane vibration to infer pitch (see, e.g., Evans, 1978b). In which direction, then, should fibers be recruited to encode changes in intensity while maintaining perceived pitch at a constant level?

Our first response, mentioned above, was to decide in favor of apicalward recruitment because psychophysical data obtained from subjects using the modiolar prosthesis indicated that increases in stimulus intensity (and therefore spread of excitation in all directions) are accompanied by increases in perceived pitch. In rereading Eddington *et al.*'s 1978 paper, though, we were reminded of their finding that pulses delivered to single electrodes in a scala-tympani array are perceived as higher in pitch when pulse duration is decreased. In contrast to the modiolar results, this finding is consistent with the evidence cited on pitch perception in the normal auditory system. That is, at short pulse durations only those fibers in the immediate vicinity of the electrode will be presented with stimuli that lie above the strength-duration curve for excitation. As duration is increased, more distant fibers are presented with stimuli above the strength-duration curve, and they will respond at a latency somewhat greater than the response latency of fibers in the immediate vicinity of the electrode. The spatial pattern of responses to long-duration pulses is broader than the spatial pattern of responses to short-duration pulses. If the central auditory system attends to the basalward edge to infer pitch, then the percept of pitch would increase with increases in pulse duration; if the apicalward edge is sensed, pitch would decrease with increases in pulse duration. Because pitch in fact decreases with increases in pulse duration, we conclude that, as in normal hearing, the falloff in driven activity at the apicalward edge signals pitch to the central auditory system. Therefore we need to revise the design presented in Fig. 2 (and

the next design to be presented in Fig. 3) so that signal intensity will be encoded by recruiting fibers in a basalward, rather than apicalward, direction.

The final component in the speech processor presented in Fig. 2 is the electrode array itself. To provide an optimistic estimate of the number of electrodes that would be required in this design for transmission of intelligible speech, we will cite intelligibility scores obtained in experiments using conventional channel vocoders and normal-hearing subjects. These scores reflect not only the performance of the vocoder systems under test but also the perceptual capabilities of the subjects. Because inputs presented to the central auditory system with cochlear prostheses are likely to be at least somewhat different from the normal inputs, intelligibility scores will probably be lower (or much lower) for implant patients. With this caution in mind we note that, for normal-hearing subjects, intelligibility scores approximate 70 % when the speech signal is synthesized from the outputs of 10 bandpass channels, each coarsely quantized to one of 3 possible levels. If similar scores could be obtained for recipients of cochlear implants using 10 bandpass channels and 3-step quantizers in the design of Fig. 2, then hope for conveying intelligible speech is realistic. That is, even a 70 % score on consonants is adequate for full recognition of speech when the syntactic and grammatical constraints of spoken English are applied. However, 40 electrodes would be required to represent the outputs of the quantizers (i.e., 10 channels \times 3 steps of quantization per channel + 10 reference electrodes = 40). The number of electrodes could be reduced to 32 if the slightly degraded performance typical of 8 channel vocoders could be tolerated. Also, the number of quantization steps could be reduced from 3 to 2, and the resulting deficit partly recovered by encoding intensity both by electrode selection and stimulus magnitude (or duration). The cost of this last change would be in a less-precise encoding of formant frequency because the spatial resolution of edge coding for pitch would suffer. With these reductions, though, the total number of required electrodes drops to 24. This number is only somewhat greater than Merzenich's estimate of independent stimulus control of up to 15 sectors of the auditory nerve with bipolar electrodes placed in the scala tympani.

Although the speech processor described in the preceding paragraphs represents a reasonable approach to the encoding problem for auditory prostheses, other approaches that "work harder" for the patient may have a greater probability of success. One such approach is illustrated in Fig. 3. Here the speech analyzer extracts the frequencies and amplitudes of the formants from the input waveform. These parameters are then encoded by selection of electrodes to receive stimuli in each time frame of ongoing input. For example, the set of electrodes selected for encoding of the third formant during one time frame is shown in the upper part of the array in Fig. 3, and is labeled F_3, A_3 . The frequency of the formant is used to select the reference electrode and the amplitude of the formant is used to determine the distance between the reference electrode and the active electrode. As in the channel-vocoder approach of the previous design, the formant amplitude is encoded by the spatial extent of excitation in the basalward direction (not apicalward, as indicated in Fig. 3). The differences between this design and the previous design are that (1) formant parameters are directly encoded in the present design, while only the

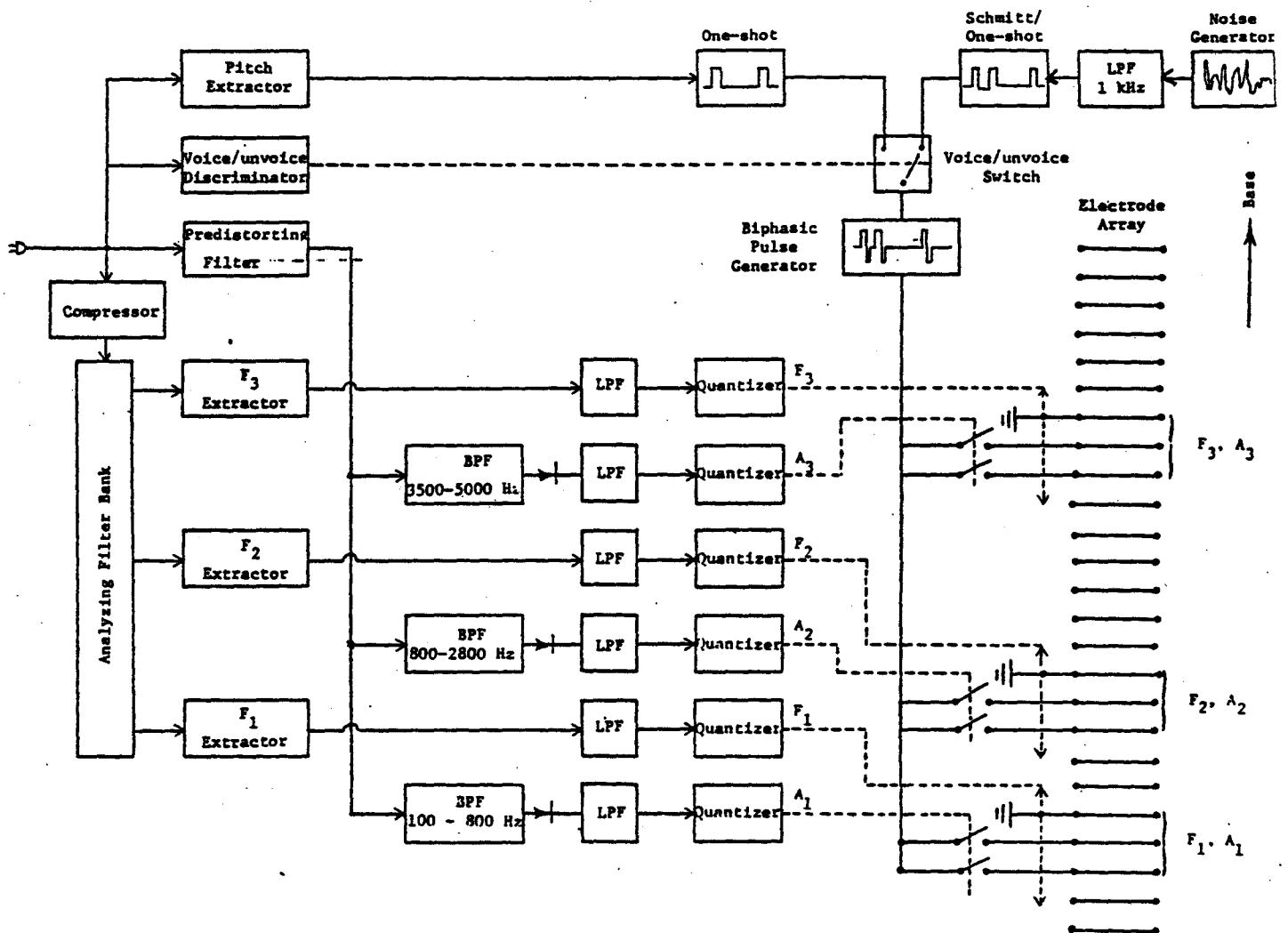


Figure 3. Formant vocoder approach as applied to auditory prostheses.

short-time spectrum is encoded in the first design and (2) the locations of the reference electrodes are not fixed in the present design, while they are in the first design. The first difference may represent a compelling advantage inasmuch as implant patients may be unable to perceive peaks in the short-time spectrum corresponding to the formant frequencies. In effect the formant "peak picker" of the second design enhances the signal-to-noise ratio for transmission of speech parameters essential to intelligibility. The second difference also represents an advantage of the second design in that "floating" reference electrodes allow for more precise encoding of formant frequencies. Finally, as developed in some detail in our original proposal, the formant-vocoder approach of the second design would also permit a further reduction in the number of electrodes required to encode the speech parameters. If the encoded stimuli presented by this processor can be fully decoded by the user's central auditory system, then it is not unreasonable to expect that intelligible transmission of speech could be achieved with an array of 15 to 20 electrodes implanted in the scala tympani.

C. References

- Bilger, R. D., Evaluation of subjects presently fitted with implanted auditory prostheses, Ann. Otol. Rhinol. Laryngol., 86, Suppl. 38 (1977) 1-176.
- Chouard, C. H. and MacLeod, P., Implantation of multiple intracochlear electrodes for rehabilitation of total deafness: preliminary report, Laryngoscope, 86 (1976) 1743-1751.
- Clark, G. M. and Tong, Y. T., A multiple-channel cochlear implant: A summary of results for two patients, Arch. Otolaryngol., 108 (1982) 214-217.
- Clark, G. M., Tong, Y. C., Martin, L. F. A. and Busby, P. A., A multiple-channel cochlear implant, Acta Oto-laryngol., 91 (1981) 173-175.
- Diller, N., Spillmann, T., Fisch, U. P. and Leifer, L. J., Encoding and decoding of auditory signals in relation to human speech and its application to human cochlear implants, Audiology, 19 (1980) 146-163.
- Eddington, D. K., Dobelle, W. H., Brackmann, D. E., Mladejovsky, M. G. and Parkin, J. L., Auditory prosthesis research with multiple channel intracochlear stimulation in man, Ann. Otol. Rhinol. Laryngol., 87, Suppl. 53 (1978) 1-39.
- Eddington, D. K., Speech discrimination in deaf subjects with cochlear implants, J. Acoust. Soc. Am., 68 (1980) 885-891.
- Evans, E. F., Place and time coding of frequency in the peripheral auditory system: Some physiological pros and cons, Audiology, 17 (1978) 369-420.
- Flanagan, J. L., Speech Analysis, Synthesis and Perception, Springer-Verlag, Berlin, 1972, 444 pp.
- Hochmair, E. S., Hochmair-Desoyer, I. J. and Burian, K., Investigations towards an artificial cochlea, Int. J. Artif. Organs, 2 (1979) 255-261.
- House, W. F. and Edgerton, B. J., A multiple-electrode cochlear implant, Ann. Otol. Rhinol. Laryngol., 91, Suppl. 91 (1982) 104-116.
- House, W. F. and Urban, J., Long term results of electrode implantation and electronic stimulation of the cochlea in man, Ann. Otol. Rhinol. Laryngol., 82 (1973) 504-517.
- Kiang, N. Y. S., Eddington, D. K. and Delgutte, B., Fundamental considerations in designing auditory implants, Acta Oto-laryngol., 87 (1979) 204-218.
- Kiang, N. Y. S. and Moxon, E. C., Physiological considerations in artificial stimulation of the inner ear, Ann. Otol. Rhinol. Laryngol., 81 (1972) 714-730.

- Klinke, R. and Hartmann, R., Auditory prosthesis: Basic physiology, Proc. XIIth ORL World Congr., Budapest, Hungary, 1981, pp. 509-513.
- Merzenich, M. M., Michelson, R. P., Pettit, R. C., Schindler, R. A. and Reid, M., Neural encoding of sound sensation evoked by electrical stimulation of the acoustic nerve, Ann. Otol. Rhinol. Laryngol., 82 (1973) 486-503.
- Mladejovsky, M. G., Eddington, D. K., Dobbelle, W. H. and Brackman, D. E., Artificial hearing for the deaf by cochlear stimulation: Pitch modulation and some parametric thresholds, Trans. Am. Soc. Artif. Int. Organs, 21 (1975) 1-6.
- Parkins, C. W., Cochlear implant: A sensory prosthesis frontier, IEEE EMB Magazine, June (1983) 18-27.
- Sachs, M. B. and Young, E. D., Effects of nonlinearities on speech encoding in the auditory nerve, J. Acoust. Soc. Am., 68 (1980) 858-875.
- Shepherd, R. K., Clark, G. M., Black, R. C. and Patrick, J. F., The histopathological effects of chronic electrical stimulation of the cat cochlea, J. Laryngol. Otol., 97 (1983) 333-341.
- Simmons, F. B., Electrical stimulation of the auditory nerve in man, Arch. Otolaryngol., 84 (1966) 2-54.
- Simmons, F. B., Walker, M. G., Mathews, R. G. and White, R. L., Percepts and discrimination by auditory nerve stimulation: A summary of results and some proposals for nerve viability evaluation. In D. L. McPherson and M. S. Davis (Eds.), Advances in Prosthetic Devices for the Deaf: A Technical Workshop, National Technical Institute for the Deaf, Rochester Institute of Technology, Rochester, NY, 1979b, pp. 271-274.
- Tong, Y. C., Black, R. C., Clark, G. M., Forster, I. C., Millar, J. B., O'Loughlin, B. J. and Patrick, J. F., A preliminary report on a multiple-channel cochlear implant operation, J. Laryngol. Otol., 93 (1979) 679-695.
- Tong, Y. C., Clark, G. M., Blamey, P. J., Busby, P. A. and Dowell, R. C., Psychophysical studies for two multiple-channel cochlear implant patients, J. Acoust. Soc. Am., 71 (1982) 153-160.
- Tong, Y. C., Clark, G. M., Seligman, P. M. and Patrick, J. F., Speech processing for a multiple-electrode cochlear implant hearing prosthesis, J. Acoust. Soc. Am., 63 (1980) 1897.
- Tong, Y. C., Dowell, R. C., Blamey, P. J. and Clark, G. M., Two-component hearing sensations produced by two-electrode stimulation in the cochlea of a deaf patient, Science, 219 (1983) 993-994.
- Walsh, S. M. and Leake-Jones, P. A., Chronic electrical stimulation of auditory nerve in cat: Physiological and histological results, Hearing Res., 7 (1982) 281-304.

Appendix 4

Hardware and Software Design for the Autocuer,
A Speech-Analyzing Lipreading Aid for the Deaf

from

"Autocuer Project Technical Status Report," November, 7, 1984.

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1. INTRODUCTION

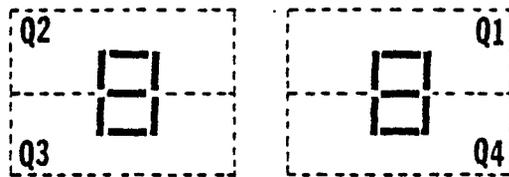
The Autocuer is a wearable microprocessor-based system that is designed to disambiguate lipreading for people who have no useful hearing; there are approximately two million such people in the United States. Based on the principles of Cued Speech, the Autocuer performs an analysis of connected speech in real time and presents the results of that analysis on a wearable eyeglasses display. As in Cued Speech, the basic display unit is a consonant-vowel (CV) syllable. Figure 1 shows the consonant and vowel groups that are used in the Autocuer. Note that the groups, particularly the consonant groups, have been selected to reduce the difficulty of the speech analysis that must be done by the analyzer in the Autocuer. For example, all of the nasal sounds are placed in the same consonant group. This means that the analyzer only needs to find that a nasal is present and does not need to make classification decisions among nasals. Similarly, four of the five unvoiced fricatives are placed in the same consonant group since all four are different from each other on the lips.

2. HARDWARE

2.1 Hardware Overview

The Autocuer hardware consists of a single-chip audio filterbank feeding speech data via an analog-digital converter to a sixteen-bit microprocessor which performs the speech analysis and drives the eyeglasses display. The display is a heads-up type display which produces a magnified image of light-emitting diode (LED) segments which are used to form one of nine consonant group symbols at one of four vowel group locations (quadrants). The Autocuer wearer centers the display about the mouth of the

THE AUTOCUER PROCESSES CONSONANT VOWEL COMBINATIONS OCCURRING IN SPEECH INTO VISUAL CUES WHICH ARE DISPLAYED ON TWO SEVEN SEGMENT LED'S. THIS LED DISPLAY IS DIVIDED INTO FOUR QUADRANTS, i.e. Q₁, Q₂, Q₃ AND Q₄.



EACH QUADRANT HAS FOUR LED SEGMENTS WHICH PRODUCE ANY ONE OF NINE DIFFERENT SYMBOLS. EACH SYMBOL REPRESENTS SPECIFIC CONSONANTS AS FOLLOWS:

SYMBOL	CONSONANT
	B, G, <i>or</i> J
⌒	K, P, <i>or</i> Z
⌒	H, S, F, <i>or unvoiced</i> TH
⌒	L, R, <i>or</i> SH
	V <i>or</i> T
⌒	M, N, <i>or</i> NG
⌒	Y <i>or</i> CH
⌒	D, ZH, <i>or voiced</i> TH
—	W, WH, <i>or no consonant</i>

THE QUADRANT IN WHICH THESE CONSONANT SYMBOLS APPEAR INDICATES THE VOWEL FOLLOWING THAT CONSONANT AS ANALYZED BY THE AUTOCUER. DIPHTHONGS ARE DISPLAYED BY MOVING FROM THE FIRST VOWEL POSITION TO THE SECOND.

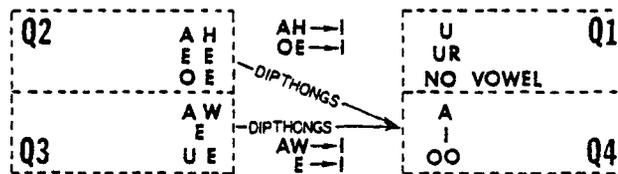


Figure 1. Autocuer Display

speaker, and the symbols appear in focus at distances appropriate for lip-reading, which is in the range of three to ten feet.

Since it must be battery powered, the Autocuer parts have been selected to minimize electrical power consumption. The total power consumption of the unit is approximately 700 milliwatts; its 5.5-ounce rechargeable battery pack will operate the unit for ten hours between recharges.

Wherever possible, the integrated circuits are of the CMOS type. The primary exception is the audio filterbank, which consumes approximately half of the total power of the Autocuer. Providing sixteen bandpass channels distributed across the speech frequencies, it is the only single-chip filterbank commercially available and is fabricated in NMOS technology. A joint development by Interstate Electronics and the Reticon Corporation, it is available from Interstate. The microprocessor is a CMOS version of the Intel 8088 and is manufactured by the Harris Corporation. The earlier version of the Autocuer used the NMOS 8088; replacing the NMOS unit with the CMOS one, a pin-compatible chip swap, reduced the total power consumption from 1.4 watts to 700 milliwatts. Three 8 Kbyte CMOS EPROMS provide the program storage, and one 8 Kbyte CMOS RAM provides for storage of speech parameters, speech phonemes, and other variables used in the analysis. The present analyzer uses 18 Kbytes of program and 6 Kbytes of data memory, so there is substantial room for program expansion if needed.

2.2 Hardware Details

2.2.1 Speech Preprocessor and Parameter Extraction

Speech is input to the Autocuer via a hearing aid microphone (Knowles Electronics type 1864). This is a directional hearing aid microphone with typically a 15 db front-to-back ratio and a 3 db front-to-side ratio. Its frequency response slopes upward by 6 db per octave from approximately 100

Hz to 4500 Hz, at which point the response falls at typically 12 db per octave. An electret type microphone, it contains an FET impedance matching transistor which requires a power supply voltage of approximately 1.25 volts, derived using a reference diode from the +5 volt regulated supply (see sheet 1 of the schematic diagrams in Appendix 1).

Because their primary spectral energy is at typically 7 KHz, four phonemes (/f, v, θ, ð/) cannot be reliably detected because they have so little energy in the range of this microphone. We would have preferred to have a microphone with usable response to 7 KHz or above, but none was available with directional characteristics and sufficiently small size. By our choice of microphone, we have opted for superior performance in noise of the directional microphone at the loss of ability to reliably detect these four sounds. Fortunately, they are quite distinctive on the lips and the effect of their usual absence from the Autocuer display on speech perception should be small.

The amplification of the speech signal from the microphone occurs in two stages. The first stage provides a gain of 101 via a Harris Corporation HA2725 bipolar operational amplifier, chosen for its good noise performance and low power consumption. The second stage has a variable gain from 26 to 76 and employs an RCA CMOS CA3420 operational amplifier; it was chosen because it offers essentially rail-to-rail signal swings and low power consumption.

The output of the second stage amplifier feeds the Interstate Electronics ASA16 spectrum analyzer. The ASA16 will operate over the voltage ranges ± 5 to ± 9 volts, so we operate it at ± 5 volts to reduce its power

consumption from typically 600 mW at ± 9 volts to 300 mW at ± 5 volts. Figure 2 shows the characteristics of the 16 bandpass filters contained in the ASAl6.

Each of the 16 channels in the unit are switched capacitor filters of the second order Butterworth type. A precision halfwave rectifier and a second order Butterworth low pass filter (25 Hz cutoff) follow the output of each of the 16 bandpass filters. An onboard multiplexer provides for sampling the output of each of the 16 filters at the selected sampling rate, which for the Autocuer is 100 times per second.

Since the ASAl6 provides the only speech parameters available to the Autocuer software, considerable care has been taken to ensure that we preserve as much of the available dynamic range per channel as possible. In particular, each of its 16 outputs has a DC offset voltage varying (with ± 5 volt power supplies) across channels from about minus 300 to plus 300 millivolts. Since it is essential that the speech analysis software be able to determine accurately the zero input signal level of each of the signal channels, the offsets must be accurately measured and compensated for each channel.

To compensate the offset voltages, we have incorporated an FET switch which is used to disable the microphone power supply (upon power up and at regular intervals during the operation of the Autocuer) to measure an offset vector and thereby determine accurately the zero input signal level for each of the 16 bandpass channels. Since each ASAl6 will have a distinct set of offsets, automating the measurement of these offsets avoids having to serialize the program ROMS for each Autocuer, and the automated conversion during operation of the Autocuer allows any offset drifts over time to be automatically compensated. In this way we can obtain a usable dynamic range

fo (Hz)	Bandwidth (Hz)	Approximate f1	Band Coverage fh
260	130	203	333
390	130	330	460
520	130	459	589
650	130	588	718
780	130	718	848
910	140	843	983
1060	160	983	1143
1220	180	1133	1313
1400	200	1303	1503
1600	220	1494	1713
1820	250	1699	1949
2070	300	1925	2225
2370	340	2206	2546
3035	1030	2563	3593
4272	1445	3610	5055
5997	2005	5077	7083

Figure 2. ASAL6 Filter Characteristics

of ~40 dB for each channel, which is the minimum dynamic range claimed by the manufacturer of the ASA16 to be available from each channel.

An 8-bit A/D converter has a dynamic range of 48 dB, so there is no loss of dynamic range by using one for sampling the ASA16 channels. A potentiometer is used to adjust the ASA 16 channel with most negative offset voltage to zero, after which the typical worst case DC offset will be approximately 600 millivolts. We use a unipolar A/D converter and set its voltage span from zero volts to 4.5 volts, so that the loss of dynamic range on the worst case channel due to offset compensation is approximately $4.5/3.9$ or 1.2 dB. In test runs to measure how well this method of compensating offset voltages preserves dynamic range, the grounded microphone condition gives a zero reference vector for the 16 channels which is within two A/D counts of zero for all channels after each offset voltage is subtracted. A maximum input signal to the ASA16, say a sinusoid peaking at ± 5 volts, produces a DC output on an individual ASA16 channel of approximately 2.2 volts, so a second RCA CA3420 operational amplifier provides a gain of 2 between the output of the ASA16 and the input to the A/D converter to boost the channel levels to cover the full voltage range of the A/D converter.

2.2.2 Digital Circuitry

The speech analysis is under the control of the Harris 80C88 microprocessor. Pin-for-pin compatible with the INTEL 8088 microprocessor, the CMOS version typically consumes one-tenth the power of the NMOS unit. With reference to sheet 2 of the schematics, the microprocessor clock (4 MHz) is derived from a 12 MHz crystal using the Harris 82C84 clock chip. The 82C84 has a one-half frequency clock called PCLK (2 MHz) which is further divided to 1 MHz for driving the ASA16 and then divided by another factor of 10,000 using an RDD104 divider chip. The output of the RDD104 provides the 100 Hz

source of interrupts for extracting a speech data sample every 10 msec from the ASA16. The interrupt source comes into the nonmaskable interrupt (NMI) pin of the 80C88; NMI was chosen over maskable interrupts because it saves about 12 microseconds per interrupt. Address buffers (U9 and U10) are 74HC373 CMOS parts, and U12 is a 74HC245 bidirectional data buffer. Eight chip selects are provided by U11, a 74HC138. All eight chip selects are used. Four are used for memory, each jumper-selectable for either EPROM or RAM. Chip selects 0, 5, 6, and 7 select the memories; as presently configured, select 0 is dedicated to the data memory; it is a Toshiba TC5565 8 Kbyte CMOS RAM. Chip selects 5, 6, and 7 are dedicated to 8 Kbyte CMOS EPROMS; we have used 27C64 units interchangeably from two different manufacturers. Chip select 1 selects a serial port to a printer for testing, and select 2 goes to the ASA16 multiplexer. Chip select 3 goes to the analog to digital converter, and chip select 4 selects the eyeglasses display driver.

The analog to digital converter is a National Semiconductor ADC0844 CMOS unit. Specified for a maximum conversion time of 40 microseconds, it has its own internal clock and typically converts in 30 microseconds from start of convert. The rather substantial reference current requirement for the ADC0844 (2.5 mA) is provided at 4.5 volts by an Intersil voltage regulator, part ICL 7663. The 74HC138, the data buffers, the ADC0844, and the memories are shown on sheet 3 of the schematics.

On sheet 4 of the schematics the display driver (a 74HC373), dropping resistors for the displays, and various other parts are illustrated. On the breadboard, we have incorporated a connector to the eyeglasses display; the cable to the display also connects the microphone contained on the eyeglasses frame to the audio amplifier on the breadboard. A small pair of 7-segment light emitting diode displays for feedback during breadboard use

(U22 and U23) also are on the breadboard. U24, a 74HC174, serves both as an address latch for the ASAL6 multiplexer and provides signals to control the FET switch which grounds the microphone power supply while measuring the ASAL6 offsets.

Sheet 5 of the schematics shows the circuitry used for deriving ± 5 volts regulated from the primary battery pack, which consists of four nickel cadmium sub-C cells connected in series. We have chosen a quick-charge type cell made by General Electric with 1.2 AH rated capacity. The recharge rate is C/3, and the recharge time is about four hours. As noted earlier, the battery pack provides sufficient power to operate the Autocuer for approximately 10 hours. The recharge jack for the battery pack is wired to preclude Autocuer operation when the battery is charging. We designed a custom regulated ± 5 volt supply for the analog circuitry because commercial units at the required current drains (30 to 40 mA) have rather poor efficiency; the best we could obtain commercially is about 50% efficient. Our design has efficiency of approximately 70% at the Autocuer current drains.

Sheet 6 of the schematics shows the serial interface to the printer used for documenting Autocuer performance in real-time operation.

3. SOFTWARE/SPEECH ANALYSIS

3.1 Speech Analysis Overview

In a global sense, the speech analysis software for the Autocuer must first make a decision about whether a given speech sound/phoneme is a consonant or a vowel, and given that decision, make a nine-way consonant classification or a four-way vowel classification, respectively. The analysis is an interrupt-driven process keyed to the 100 Hz clock. Every 10 msec an interrupt occurs and causes another set of sixteen samples of speech, one from each bandpass filter in the filterbank, to be brought in for analysis.

Each sample is a one-byte quantity representing the energy in its respective frequency band during the last 10 msec time interval. The interval of 10 msec is chosen because no significant vocal tract change occurs in less than 10 msec. The shortest phoneme-like event, say a stop burst, will never be smaller than about 20 msec, and the vast majority exceed 50 msec.

The next major step in the analysis is to segment the speech at phoneme or phoneme-like intervals; accurate segmentation is both the most important and most difficult step in the analysis that the Autocuer must perform. The present version of the segmenter performs segmentation at an accuracy of 75 to 85 percent, as compared to careful segmentation by a phonetician using speech spectrograms, when it operates on the sentences on which the analyzer was developed. Its accuracy on new speech is approximately 70 percent. The segmentation is based on broad-band energy peaks which are computed by summing the energy in the top 11 and in the bottom 5 frequency bands; both a high frequency (900-7000 Hz) and a low frequency (200-900 Hz) segmentation are performed.

After segmentation is completed, the consonant/vowel decision is made. It is based on time alignment between high and low frequency segmentation "phonemes" plus a number of energy balance and energy level measures.

Vowels are classified by taking the ratio of the first two formants at the point of maximum energy in the phoneme as marked by segmentation. Algorithms are implemented to estimate the frequency of a formant to an accuracy of 25 Hz, which is about one-sixth of the typical bandwidth of a bandpass filter in the filterbank. This is done by comparing the energy in the two bands adjacent to a band with an energy maximum and using an interpolation formula based on the shape of the filter skirts to adjust the frequency estimate. Since formant frequency is inversely proportional to vocal tract

size, use of the formant ratio rather than the formants directly is an effective way to make the analyzer less speaker dependent.

Consonants are classified by a variety of spectral and temporal measures. The order in which consonants are tested for membership in various consonant classes (fricatives, plosives, nasals, etc.) is in order of decreasing accuracy, i.e., the consonants we can classify most accurately are tested for first. This order is presently plosives, fricatives, affricatives, nasals, and liquids. Plosives are classified by the presence of the silent interval preceding the release of the burst, by the duration of the silence and that of the burst, and by the frequency of the maximum energy peak at time at which the energy peaks in the high frequency broadband as marked by high frequency segmentation.

Fricatives are classified by high/low energy balance and by the frequency of the maximum energy peak in the filterbank at the time of the maximum broadband high frequency energy peak. Since an affricative is actually a plosive run together with a fricative, the classification of affricatives is essentially a combination of the tests used to classify plosives and fricatives. For plosives, fricatives, and affricatives, the determination of whether a phoneme is voiced or unvoiced is made by computing the voice onset time, which is the time from onset of high frequency energy to the time of onset of voicing. For example, the decision of whether a plosive is voiced is made by determining whether voicing begins within 40 msec of the onset of burst energy, in which case the plosive is called voiced; if it begins over 40 msec from burst release, the plosive is called unvoiced.

Nasals are classified through the use of energy balance between certain frequency bands, between high/low frequency broadband energy balance, and a

variety of measures based on the phoneme environment in which the phoneme occurs. The analyzer presently performs reasonably well in finding and classifying phrase-initial and phrase-interior nasals, but does not yet have an effective algorithm for finding and classifying phrase-final nasals. It still needs significant work on finding and classifying liquids.

Timing of the cue strings which represent the final output of the speech analysis is under control of the same interrupt routine which gathers the raw speech data via the filterbank. The timing out of any cue is always to a precision of 10 msec, and the cue duration faithfully replicates the actual duration of the speech syllable it represents.

3.2 Software and Speech Analysis Details

The speech analysis software consists of a series of modular programs linked together by a dispatcher module. With the exception of one module, all software is written in PLM86 compiler code. The speech analysis software, per se, is approximately 18 Kbytes in length, of which all but 600 bytes are in PLM86 source code. Table 1 shows the name, indicates the function, and gives the length in bytes of each of the major modules.

With reference to Table 1, the dispatcher, DISPATCH, controls the overall flow of the program. The minimum segment of time processed is 10 msec, chosen to match the minimum time for meaningful events to occur in the vocal tract. Upon program startup, RAM is initialized, and interrupts are enabled. A count of the minimum segment of speech, MINSEGCOUNT, is tested to control the processing of speech as it is brought in by the interrupt-driven data gathering module, GATHER. Upon the receipt of an interrupt, GATHER tests for the presence of a cue to be sent to the display, and if one is ready, it activates the display subject to the previous cue having been completed. It then tests for whether time is to be reset, i.e., if data are

TABLE 1. AUTOQUER SOFTWARE MODULES

Name	Function	Size in Bytes
1. DISPATCH	Controls overall program flow and builds cues.	1765
2. GATHER	The interrupt-driven module, it gathers speech data, turns cues on and off, and controls the speech buffers.	2398
3. PKDIP	Finds all peaks and valleys in the HF and LF broadband energy buffers.	979
4. HISEGMEN	Segments HF broadband energy via energy peaks and valleys.	3710
5. LOSEGMEN	Segments LF broadband energy.	2857
6. CLASSIFY	Controls the order of calling of routines that classify phonemes.	567
7. RECOG	Contains the various routines that classify phonemes (CONSORVOW, VOWEL, FRICATIVE, NASAL, etc.)	5360
8. FREQ/FREQPEAK	Finds energy peaks in ASA16 channels and estimates vocal tract formants.	751
9. DISPAT.ASM	The assembler program. It contains initialization routines, interrupt handler, and support software for RS232 interface.	523

to be thrown away. Time is always reset upon the recognition of a phoneme or phoneme pair representing a cue. Once processing has begun, we always maintain a minimum of ten 16-byte sets of ASA16 data samples, namely 100 msec, of speech in the buffers. It is very useful in speech analysis to have access to the previous short-time speech history to ferret out phonemes such as nasals, liquids, and the silent intervals preceding stop bursts.

Once 50 msec of data are acquired, and every 50 msec thereafter, the procedure PKDIP is executed to compute all the peaks, valleys, and major slope changes in two aggregate data sets comprised of the broadband low frequency energy (the 5 low channels from the ASA16) and the broadband high frequency energy (the 11 high frequency channels). Upon completion of PKDIP, the procedure HISEGMEN which segments the high band energy is called and followed by the procedure LOSEGMEN which segments the low band energy. HISEGMEN and LOSEGMEN each compute and maintain "phoneme" tables called, respectively, PHONEM and LOPHONEM. For every pass through the real-time loop entered for each 50 msec of additional speech data, the peaks, valleys, major slope changes, and phonemes, if available, are found.

Having caught up with the available speech data in the above procedures, we enter the phoneme classification loop. There, we call a procedure COMPUTEPHONEMECOUNT which determines if one or more complete phonemes are in the phoneme table which have not yet been classified. If so, we next test for silence at the end of the phoneme about to be classified; in every case in which silence follows the phoneme to be classified, the cue representing that phoneme must be activated immediately after classification of that phoneme. Otherwise, if the phoneme classified is a consonant, its cue code and cue time must be saved until the following phoneme is classified. If the phoneme following the consonant is a vowel, then the consonant cue is combined with it. If the phoneme following is a consonant, that consonant

code and its time are saved and the previous one sent as a consonant in isolation.

The order in which classification procedures are called is contained in module CLASSIFY and begins with determining whether the phoneme is a consonant or a vowel. The procedure that implements this, CONSORVOW, tests to determine if for the high frequency under classification there exists a low frequency "phoneme" which is time-aligned to it within 60 msec. This is done by determining if the significant high frequency (HF) energy peak marking the "center" of the HF phoneme in the PHONEM table time aligns within 60 msec of a significant low frequency (LF) energy peak, for some low frequency phoneme in the LOPHONEM table. If so, if there is a minimum time overlap of 50 msec between high and low frequency phonemes, if the HF and LF peaks meet certain minimum energy criteria, and if the HF to LF energy ratio is within certain bounds, then the phoneme is called a vowel. Otherwise it is called a consonant.

For vowels, classification is done by computing the ratio of the second formant to the first formant, as determined from the ASA16 data. The frequency estimates are always taken at the time at which the energy in the HF region is a maximum; this in general will correspond to maximum signal to noise ratio and is representative of the time at which formant estimates for vowels in isolation are rather stable.

Since diphthongs are in effect two vowels run together, the routine VOWEL which classifies vowels also classifies diphthongs. To determine if a vowel is a diphthong, we first examine the HF phoneme to determine if it has two energy peaks each of which has been marked by HISEGMENT as significant. Significant peaks are marked by HISEGMENT and LOSEGMENT based upon the ratio between a peak and an adjacent valley or major slope change. For most

diphthongs, there will be two significant energy peaks characterized by well-defined valleys (or slope changes) to the left and to the right of a left peak and a right peak, respectively, with only a minor valley separating the two. The segmenters always mark both the left peak and the right peak of such phonemes; and the existence of a doubly-marked HIPHONEM table entry causes entry to a diphthong search. Subject to certain durational constraints, a vowel is subdivided into two vowels -- a diphthong -- based upon the behavior of the formants at the two energy peaks. Because diphthongs often do not reach the target formants for the second vowel in a diphthong, the vowel classification is modified to accept somewhat different formants for the second half of a diphthong than for the vowel corresponding to that second half when that vowel is spoken in isolation.

Two procedures are used to process the ASA16 data for classifying both vowels and consonants. These procedures are FREOPEAK and FREQ. FREOPEAK is called to determine the channel numbers at which energy maxima exist for the phoneme under classification at the HF broadband peak energy time. The procedure FREQ is then called to estimate the actual frequency represented by a given energy peak. For vowels, we determine the frequency of the first two significant energy peaks; these are called Formant 1 and Formant 2. The vowel classification is based upon Formant 2 divided by Formant 1, after appropriate scaling to preserve significance. For consonants, examination of the filterbank data at the broadband energy peak depends upon the consonant class. For both fricatives and stops, we identify the maximum energy peak and use the frequency of the maximum energy peak for the actual classification decision. Stops and fricatives are both noise-like and therefore have rather broad energy peaks, and the use of the maximum energy peak works quite well for most of the phonemes in these two classes. For the four fricatives noted earlier whose primary spectral peaks are at about 7 KHz, we

must use secondary energy peaks which are within the passband of the microphone. For example, the fricatives /f/ and /v/ have a secondary peak at approximately 2 KHz; if a fricative-like region with low overall energy is found, the major energy peak at 2 KHz provides a primary input to the /f, v/ classification decision.

Nasals are classified based upon the energy ratios of the broadband HF and LF data sets and certain of the individual ASA16 low frequency channels, particularly frequency bands 1 through 3. Algorithms are incorporated to detect and thereby classify initial nasals and phrase-interior nasals. We have not yet found a reliable way to detect phrase-final nasals, because their low energy is not clearly distinguishable from the gradual trailing off of energy as the articulation of a vowel is completed.

The ability to accurately estimate the frequency of a vocal tract resonance is very important to accurate classification, particularly for vowels and vowel-like sounds. Consequently, we have incorporated in the procedure FREQ an algorithm to estimate the actual frequency of a given energy peak with much better resolution than is available in bandwidth of the filters themselves. We examine the filter channel to the left and to the right of a filter channel with a peak. If the energy in the channel adjacent on the left to the peak channel is greater than the energy in the next channel adjacent on the right of the peak, the true frequency of the input source is to the left of the center of the peak channel and similarly for the reverse case. We use an interpolation formula based upon the bandwidth of each filter and the shape of the filter skirts (recall that we have second order Butterworth bandpass filters) to make the adjustment to the actual frequency of the source. Throughout the critical speech frequency range from 400 Hz to 2400 Hz, tests with sinusoidal inputs verify that we

can accurately estimate the frequency of the source to within 25 Hz, much better than the bandwidth of the filters themselves.

The "liquid" phonemes /l, r, w/ are fairly similar to the nasals in that they usually do not have sufficiently well-defined energy peaks to reliably segment them using our segmentation schema. We have found that the sound /w/ is particularly difficult to segment and therefore have included it with the no-consonant category so that a segmentation decision is not necessary. Classification routines need to be completed for the three phonemes /l, r, y/.

4. COMMERCIALIZATION AND PATENT STATUS

The exclusive license agreement between Research Triangle Institute, Gallaudet College, and Telesensory Systems, Inc. is based upon the Autocuer system patent (pending) and the Autocuer display patent (U.S. Patent 441443).

Most of the parts necessary to build 24 wearable field test Autocuers have been procured. Those yet to be purchased in quantity are the following:

1. ASA16 spectrum analyzers
2. Eyeglass displays
3. Battery chargers and battery packs
4. Autocuer cases
5. Printed circuit boards for the wearable Autocuers. (These are to be provided by TSI)

It is with pleasure we note that the Harris CMOS 8088 microprocessors for which we have been waiting so long arrived late in October 1984 at RTI, in sufficient quantity to build the necessary units for the field test.

5. TRAINING AND FIELD TEST STATUS

Gallaudet College continues to have the responsibility for pretraining and actual execution of the field test. RTI has the responsibility for maintaining the field test units during the actual field test. Staff at Gallaudet are now ready to begin the selection of field test subjects and pretraining of these subjects subject to the extension of the Gallaudet subcontract to RTI.

APPENDIX 1

AUTOCUER PARTS LIST
AND SCHEMATICS

AUTOCUER PARTS LIST

LOCATOR DESIGNATION	PART NUMBER	MANUFACTURER	FUNCTION
U1	HA-2725	Harris	OP AMP
U2, U4	CA3420AE	RCA	OP AMP
U3	ASA-16	Interstate Electronics Corp.	Speech Preprocessor
U5	32C84A	Harris	Clock Generator/Driver
U6	MM74PC74J	Mitel	Dual D Flip-Flop
U7	RDD 104	LSI	Four Decade Divider
U8	80C88	Harris	Microprocessor
U9, U10, & U19	MC 74HC373H	National	Octal 3-State Latch
U11	MC 74HC138N	National	Demultiplexer
U12	MC 74HC254N	National	Octal 3-State Bus Transceiver
U13	ICL 7663CPA	Intersil	Voltage Regulator

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U14	ADC 0844	National	A/D Converter with 4-Channel Multiplexer
U15	TC5565 PL-15	Toshiba	RAM
U16, U17, & U18	D2764	Intel	EPROM
U20	14-Pin DIP Component Platform Containing R20-R26		
U21	14-Pin DIP Component Platform Containing R27-R33		
U22 & U23	MAN 74A	Fairchild	Seven Segment LED Display
U24	MM 74HC174N	National	Hex D Flip-Flop
U25	MC 74HC02N	National	Quad 2-Input NOR Gate
U26	MC 74HC00N	National	Quad 2-Input NAND Gate
U27 & U28	750-101-R22K	CIS Corp.	Pull-Up Resistor Network
U29	CD4047AE	RCA	Flip-Flop
U30	14-Pin DIP Platform Containing T1, Q3, and Q4		
U31	ICL 7663CPA	Intersil	Voltage Regulator

A-4-21

R1, R2, R3, R8, R36, & R39	RN55D-10K		1/8W, 1% Resistor
R4, R5, & R16	RN55D-100K		1/8W, 1% Resistor
R6, R11, R20, R21, R22, R23, R24, R25, & R26	RN55D-1K		1/8W, 1% Resistor
R7, R37, & R85	RN55D-6.8K		1/8W, 1% Resistor
R9 & R19	960-20 (100K)	VRN	20 Turn Trimpot
R10 & R437	RN55D-75K		1/8W, 1% Resistor
R12	960-20 (2K)	VRN	20 Turn Trimpot
R13 & R14	RN55D-39.2K		1/8W, 1% Resistor
R15	960-20 (20K)	VRN	20 Turn Trimpot
R17	RN55D-8.06K		1/8W, 1% Resistor
R18	RN55D-7.5K		1/8W, 1% Resistor
R27, R28, R29, R30, R31, R32, & R33	RN55D-402		1/8W, 1% Resistor

A-4-22

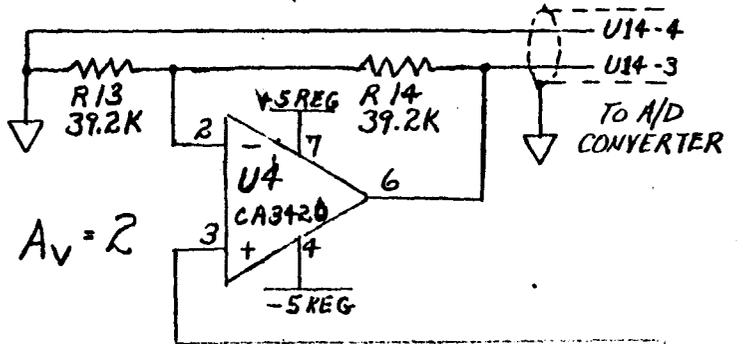
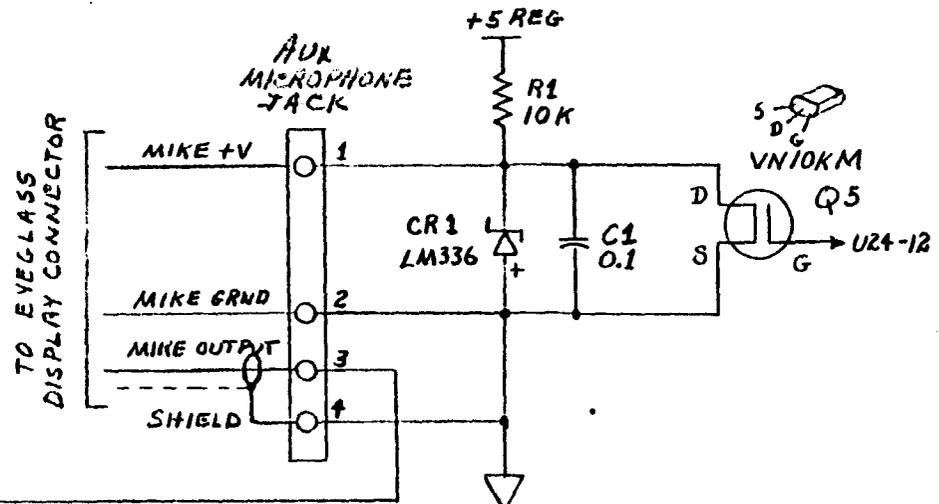
R38	RN55D-25.5K		1/8W, 1% Resistor
R40	960-20(1K)	VRN	20 Turn Trimpot
C1, C2, C3, C4, C5, C6, & C8	8121M050 X7 R (0.1 F)	Sprague	Capacitor
C7	47pF		5% Capacitor
C9	100pF		5% Capacitor
C10 & C11	10 F		6V Electrolytic Capacitor
C12, C22, & C23	IC25254474M050B 0.47 F	Sprague	Capacitor
C13	1.0 F		6V Electrolytic Capacitor
C14 & C15	47pF		Capacitor
C16	100pF		5% Capacitor
C17 & C18	33 F		10V Electrolytic Capacitor
C19, C20, & C21	100 F		10V Electrolytic Capacitor

A-4-23

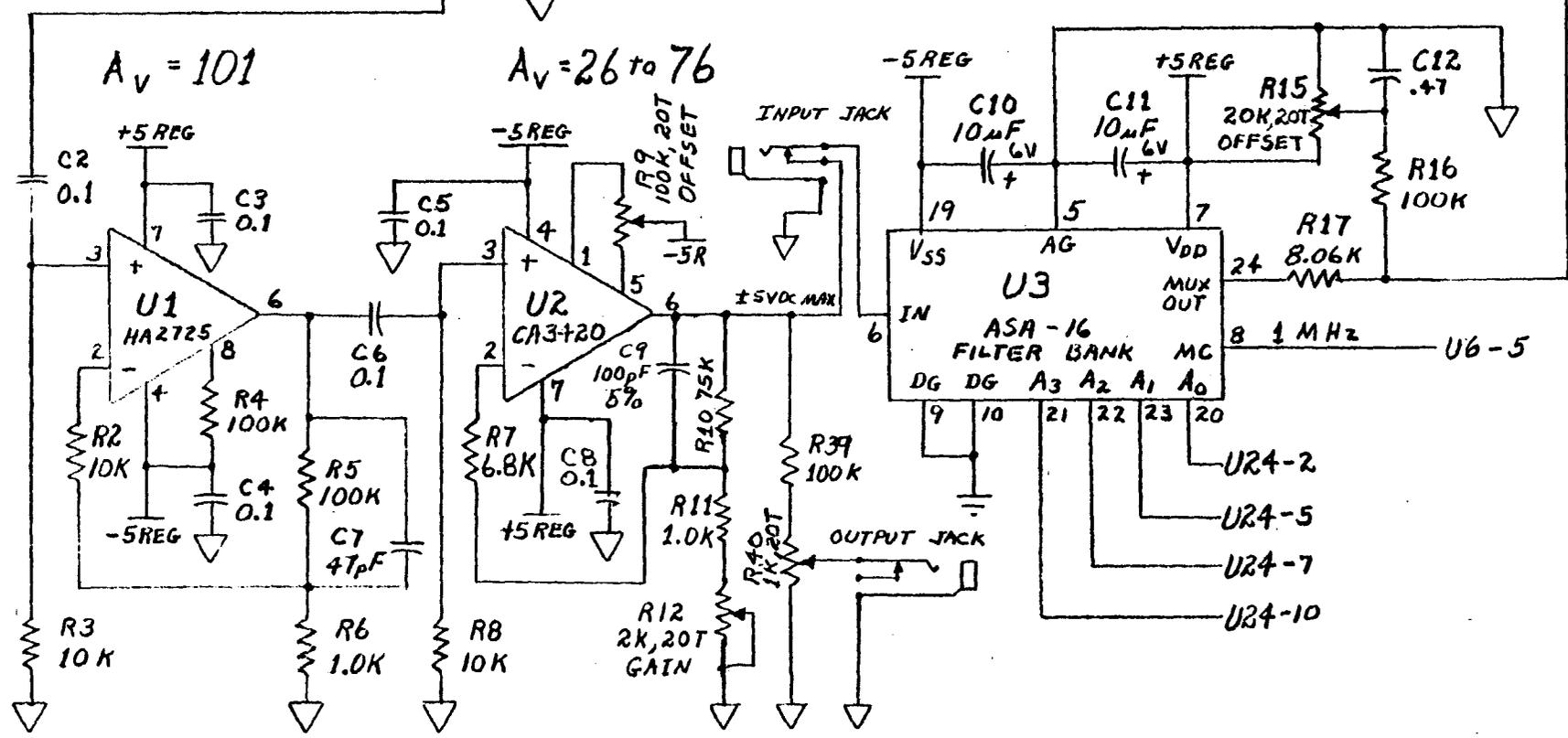
CR1	LM330	National	2.5V Voltage Reference
XAL	12 MHz		Crystal
D1, D2, & D3	1N914		Diode
D4, D5, D6, & D7	1N5817		Diode
S1			Momentary Switch, N.O.
S2	TT-23N-2T	ALCO	Switch, DPDT
S3	TT-13A-2T	ALCO	Switch, SPST
Eyeglass Display Connector	FR-125-6	Microtec, Inc.	Breadboard Connector
	FP-125-1	Microtec, Inc.	Eyeglass Display Connector
Q1 & Q2	2N2222		Transistor
Q3 & Q4	VN 10KM	Siliconix	Transistor
T1	Custom Wound Transformer. #32 Wire on Ferroxcube 266CT12513BT Core		
L1	Custom Wound Inductor. 13 Turns #26 Wire On Micrometals T25-26 Core		
Input Jack & Output Jack	142A	Switchcraft	Miniature Microphone Jack

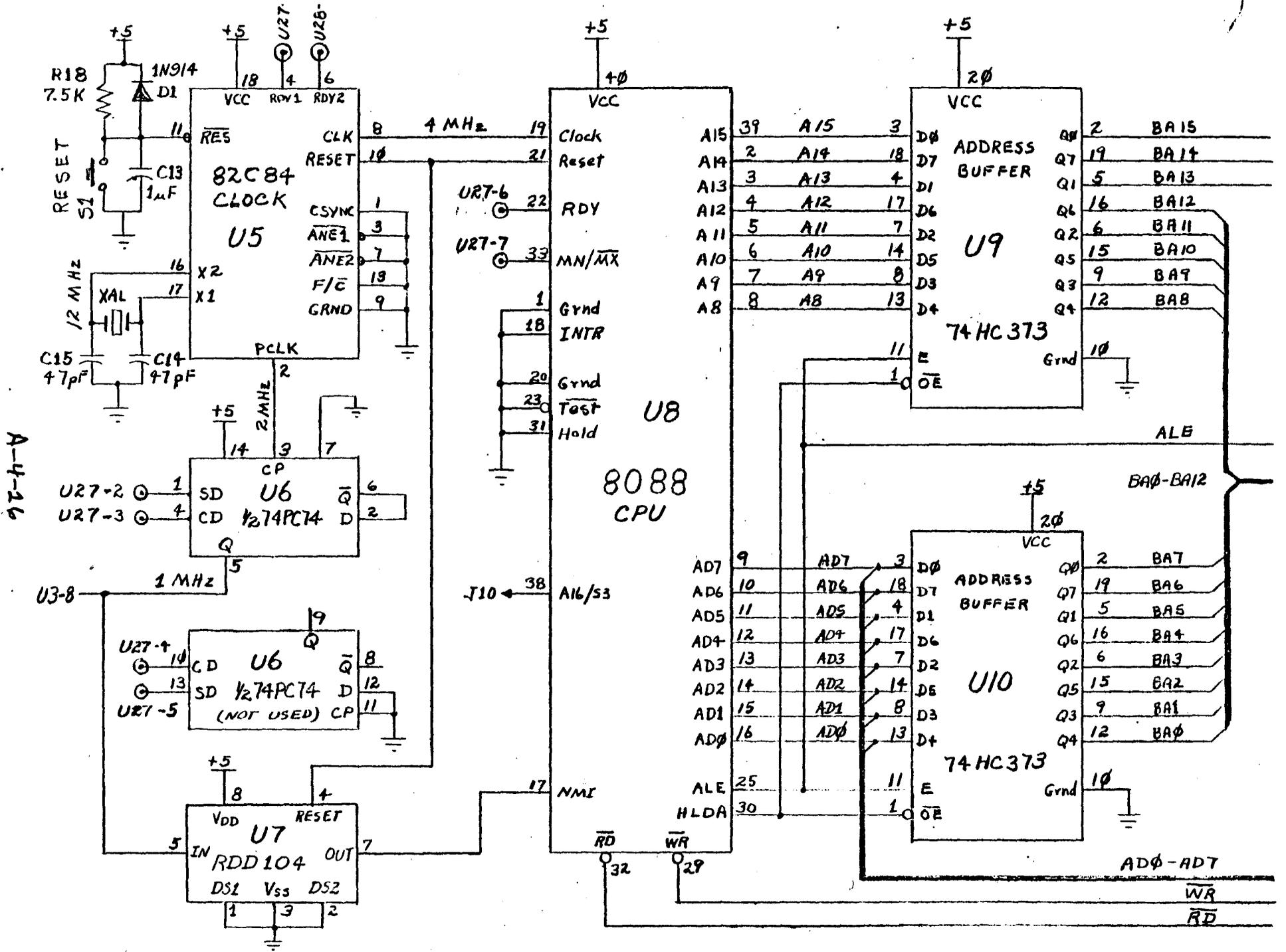
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AUTOCUER
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PHIL



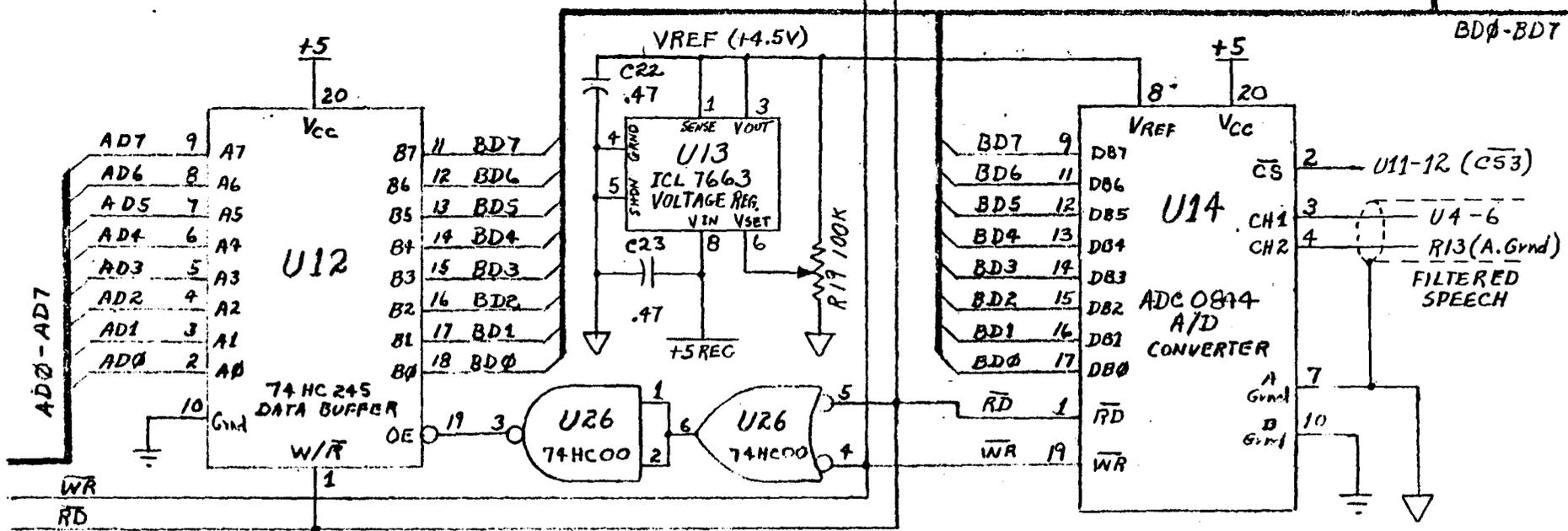
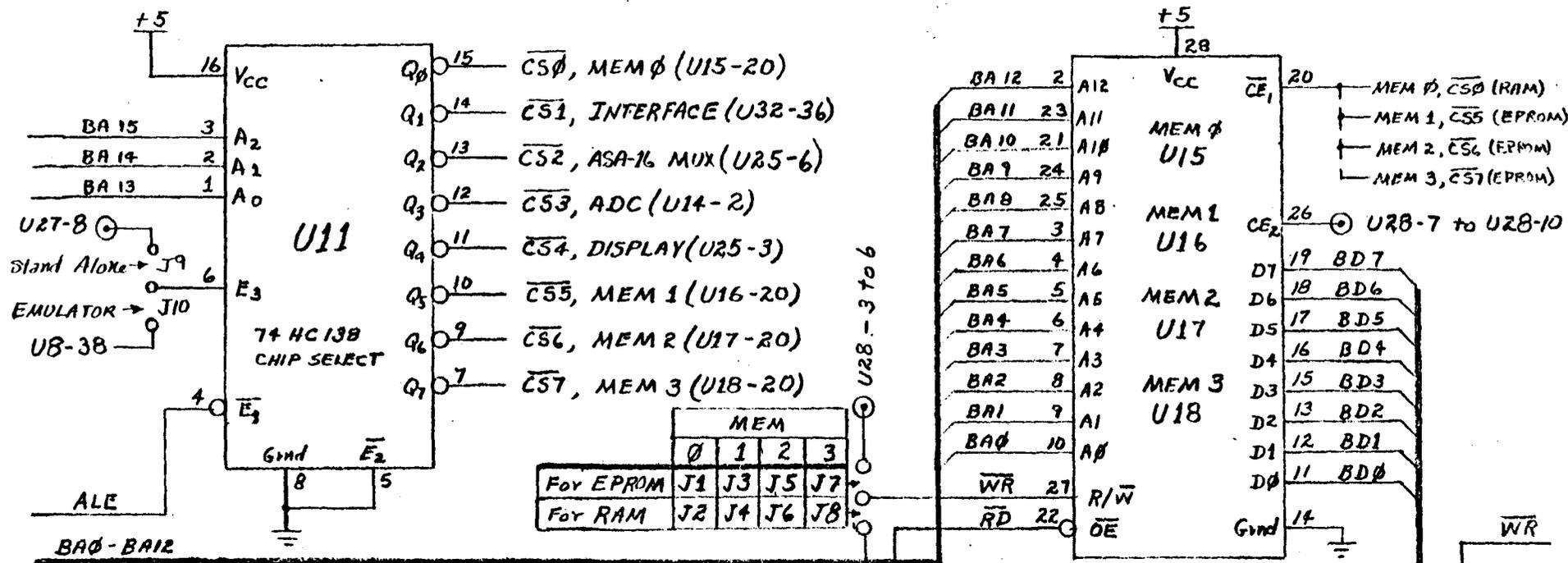
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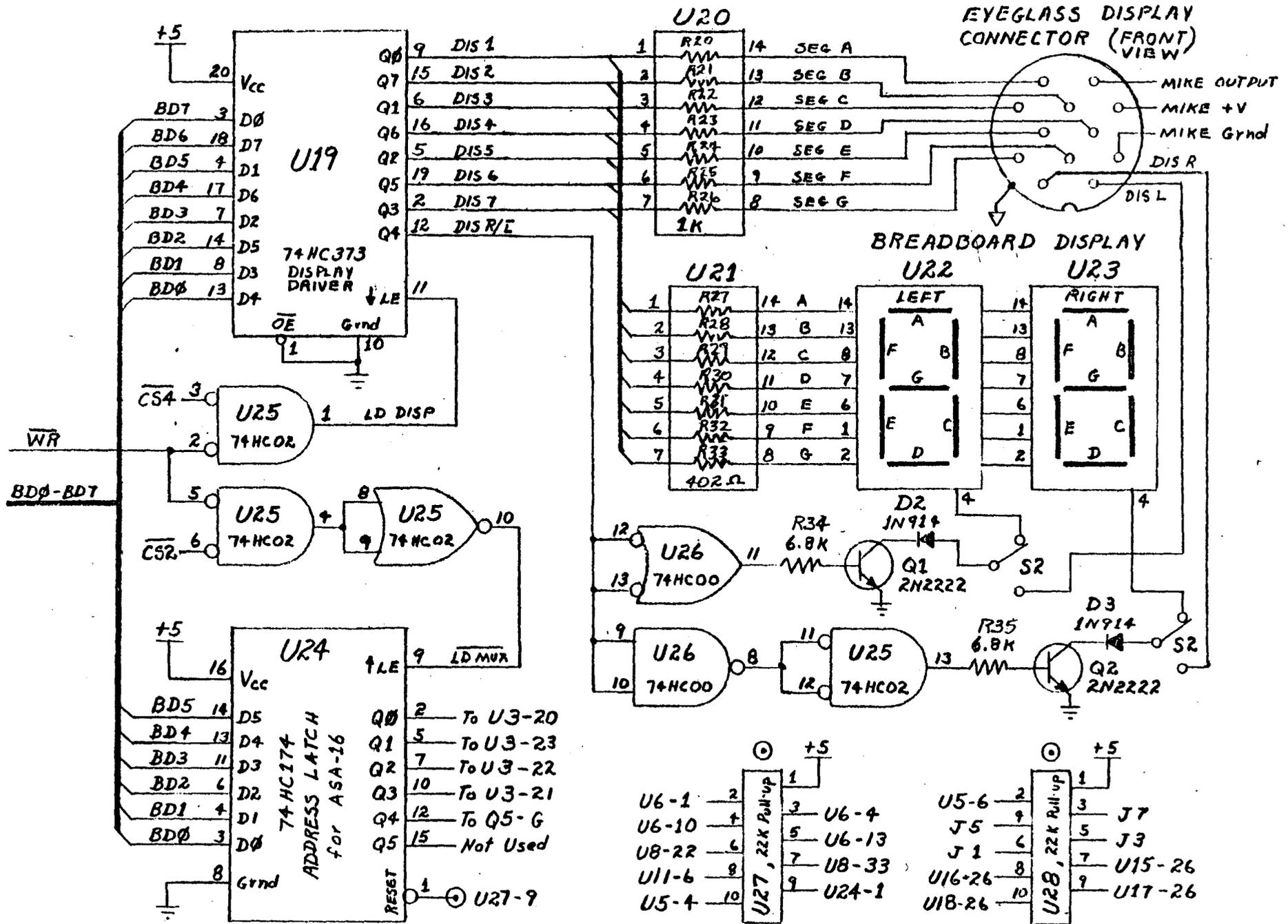


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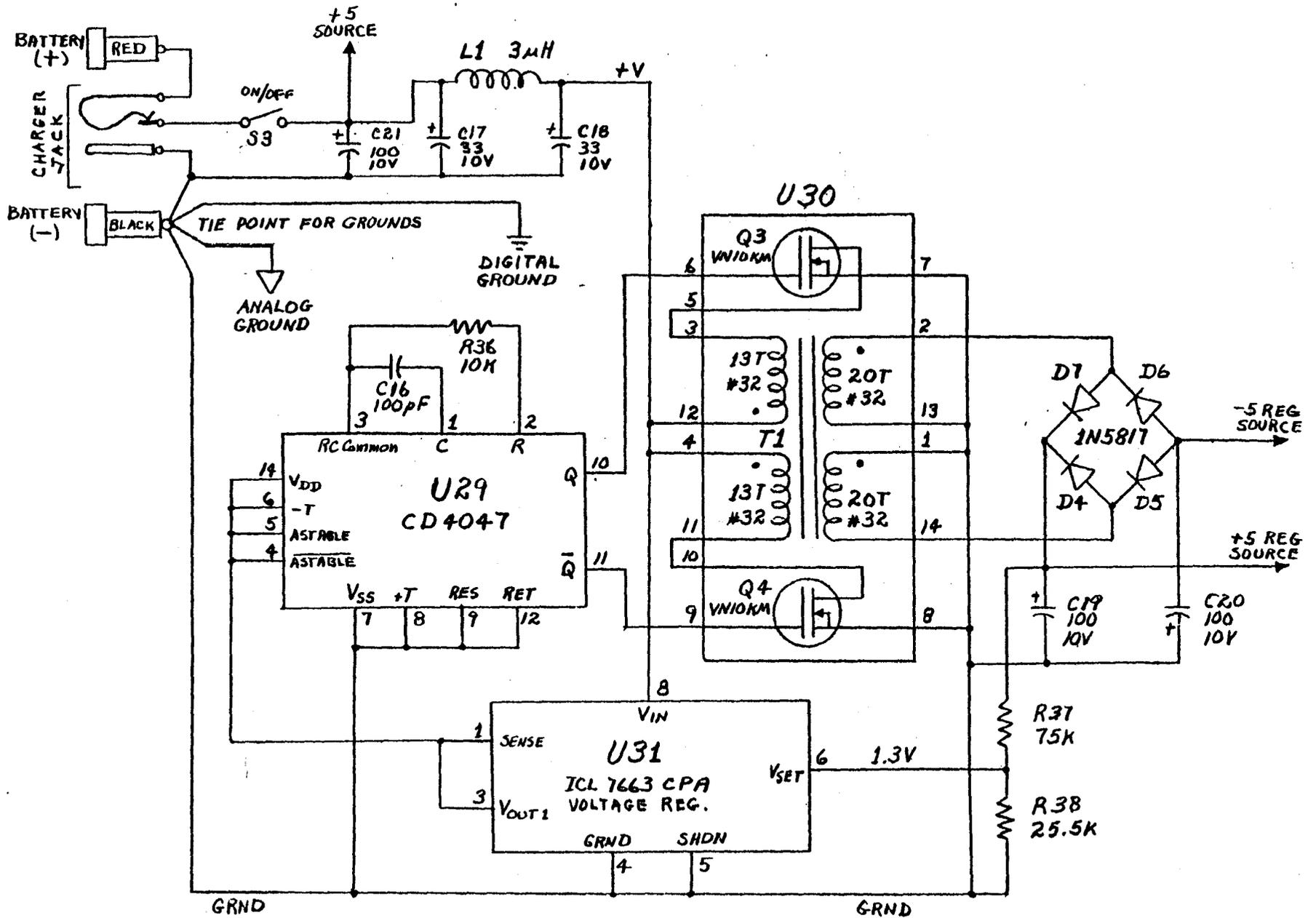
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A-4-18

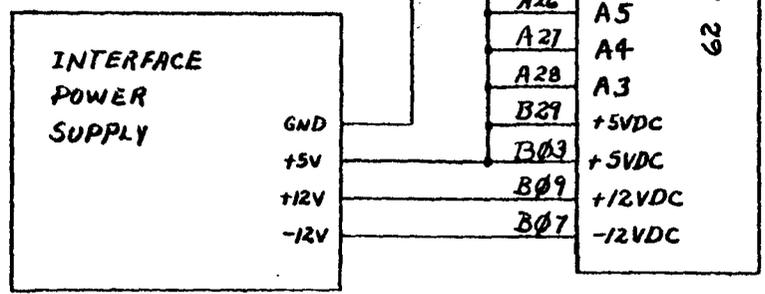
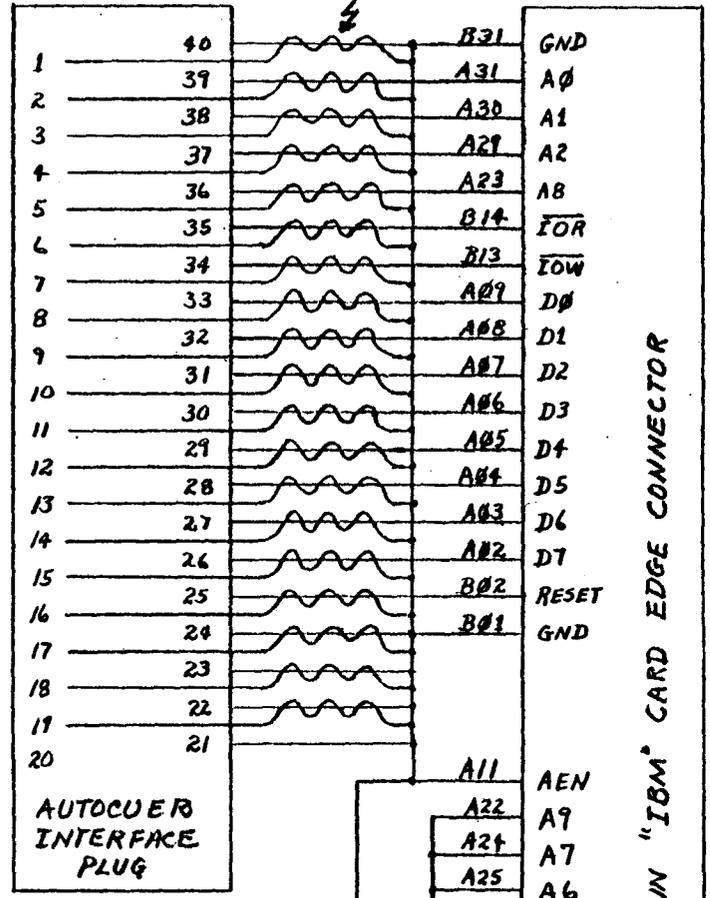
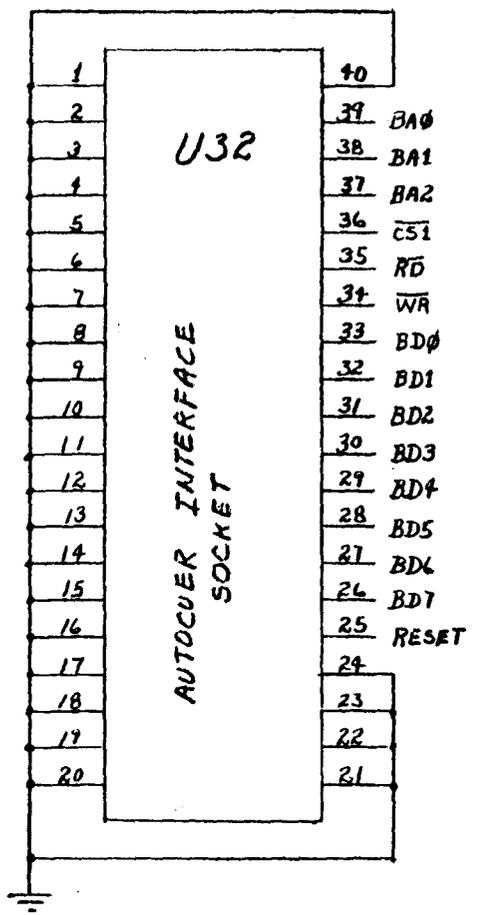


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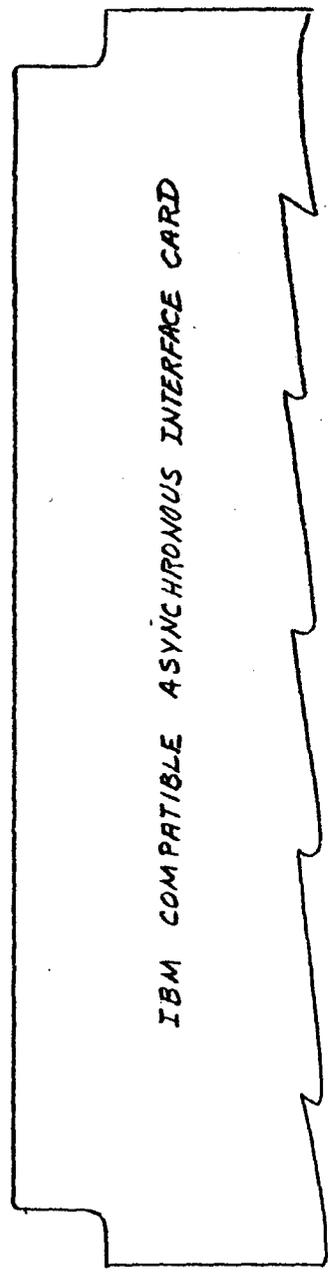


18" 40 CONDUCTOR, TWIST-FLAT, RIBBON CABLE

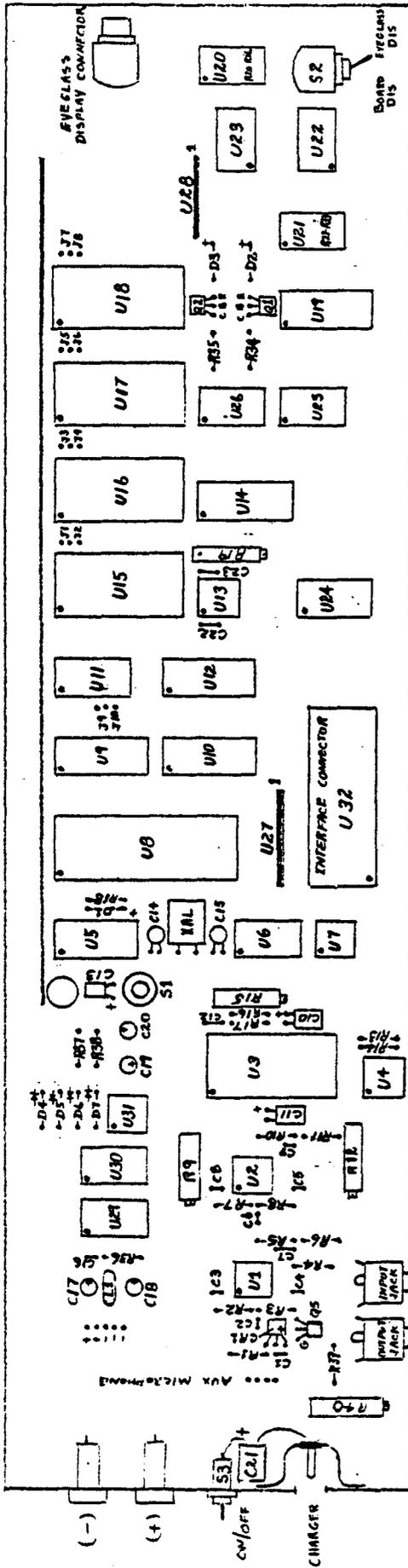
A-4-30



62 PIN "IBM" CARD EDGE CONNECTOR



→ AUTOCUER INTERFACE BOX



AUTOQUER BREADBOARD
 PARIS LOCATER
 W/7/84 Phil

Appendix 5

General Description of the Research Triangle Institute

RESEARCH TRIANGLE INSTITUTE

General Qualifications

Research Triangle Institute (RTI) is a not-for-profit contract research organization centrally located on a 180-acre campus in the Research Triangle Park between Raleigh, Durham, and Chapel Hill, North Carolina. The Institute was incorporated as a separate entity in 1958 by The University of North Carolina at Chapel Hill, Duke University at Durham, and North Carolina State University at Raleigh. The Institute also maintains a close affiliation with North Carolina Central University at Durham.

Institute research is performed both in the United States and abroad under contract with Federal, State, and local governments, public service agencies, and industry clients ranging from small companies to national corporations. RTI's operations are divided into major groups covering social sciences; statistical sciences; chemistry and life sciences; and energy, engineering, and environmental sciences. Centers, divisions, and offices within the groups conduct research in a wide variety of subject areas. RTI programs combine extensive experience, capabilities, and resources in management, information, and systems analyses and in physical, engineering, and laboratory research, thus reflecting an unusual breadth of research capability and scientific interest.

The total staff of the Institute numbers over 1,200 persons. Approximately 65 percent of the full-time staff are professionally trained research personnel. Of these, 25 percent hold Doctoral degrees, and another 35 percent hold Master's degrees. The backgrounds of the professional staff cover more than 80 degree fields and provide a firm base for conducting interdisciplinary studies. Composition of the research staff is:

- 160 Chemists and Pharmacologists (74 with graduate degrees): analytical, organic, toxicology, and polymer
- 120 Engineers and Physicists (98 with graduate degrees): aeronautical, agricultural, biomedical, chemical, civil, electronic, electrical, solid state, industrial, sanitary, environmental, mechanical, mining, nuclear, petroleum, and transportation
- 110 Mathematicians, Statisticians, and Computer Scientists: (60 with graduate degrees)
- 60 Environmental Scientists (30 with graduate degrees): industrial hygiene, epidemiology, biology, geology, meteorology, ecology, and public health

• 225 Social Scientists (110 with graduate degrees): economists, sociologists, psychologists, political scientists, and social psychologists

Other professional staff members encompass a wide variety of additional disciplines, including agricultural sciences, education, business administration, and the humanities.

The availability of extensive library facilities at the neighboring universities is one of the Institute's significant research resources.

The combined university libraries have the largest collection in the South, with over 6 million volumes and over 17,000 journals and periodicals cross-catalogued and shared since 1934. The combined scientific periodical collections are indexed in "The North Carolina Union List of Scientific Serials," which is updated by computer processing. Access to these resources by loan or by copying services is facilitated by daily truck service between the universities and the Institute. RTI staff members are granted full library privileges.

Another valuable research resource is the North Carolina Science and Technology Research Center (STRC) which is located adjacent to RTI in the Research Triangle Park. STRC provides computerized information searches of more than 100 different bases including Chemical Abstracts, Engineering Index, and most of the important Government data bases. STRC has its own collection of more than 400,000 documents of unpublished reports, journal articles, books and conference papers covering many scientific fields of interest.

For its data processing activities RTI uses its own in-house capabilities, the adjacent Triangle Universities Computation Center (TUCC). RTI's in-house computers include a PDP-11/60, a PDP-8E, an EAI-380, a Nova 820, a Nova 2/10, two Eclipse S/140s, a dual Data General Eclipse S/230, and a Zendex microcomputer development system.

The TUCC facility is comparatively large, with hardware consisting of an Amdahl 470 V/8 and dual IBM 370/165 central processing units, over 16 million bytes of memory, substantial tape and disk capabilities, and extensive teleprocessing facilities via both dial-up and dedicated ports. The TUCC software includes all the standard IBM compilers, loaders, and utility programs, plus an extensive array of compilers, statistical packages, and other products from various sources.

RTI interfaces with TUCC via four medium-speed terminals: a Data General S/140, a Harris 1610, a Datapoint 4520, and a Datapoint 1100. The Datapoint can interface with IBM, CDC, UNIVAC, OSI, COMNET, and DEC systems. RTI also has daily traffic between RTI and IBM systems at Optimum Systems Incorporated (OSI), Computer Network Corporation (COMNET), The Federal Government's Park-lawn Computer Center, and EPA's National Computer Center (NCC); there is occasional traffic with the Federal Emergency Management Agency (FEMA) Computing Center in Maryland.

Facilities that are available to RTI but located on-campus at the local Universities include a VAX-11/780 at North Carolina State University, two VAX-11/780s at The University of North Carolina at Chapel Hill, and various minicomputers at all three universities.

RTI's Communicating Word Processor (IBM 6/430) can communicate bi-directionally with TUCC, other computers, and other word processors bi-synchronously via 1200, 2000, and 2400 baud dial-up lines, and can output the text or data on medium-speed ink jet printers or phototypesetter.

The Institute's cognizant administrative contract office is Defense Contract Administration Services Management Area, Atlanta, 805 Walker Street, Marietta, GA 30060, Code: DCRA-DAB, Attention: Janice P. Watson, Telephone (404) 429-6017.

Accounting and purchasing practices may be confirmed by contacting the HHS Audit Agency, Post Office Box 27443, Raleigh NC 27611, Attention: Phil Maddox, Telephone (919) 755-4226.

RTI's top secret facility clearance held since July 5, 1961, may be verified through Defense Contract Administration Services Region, Atlanta, Directorate of Industrial Security, 805 Walker Street, Marietta, GA 30060, Telephone (404) 429-6000.

C. Collaborative Arrangements

We have made arrangements to continue our ongoing collaborations with the cochlear implant teams at the University of California at San Francisco (UCSF) and at the Duke University Medical Center (DUMC). In addition, we have a collaborative agreement with Storz Instrument Company of St. Louis to evaluate single channel coding strategies. All of these collaborations are fully described in section II.F of this proposal. Formal letters of collaboration from the groups at UCSF and DUMC are reproduced on the following pages.

**John C. & Edw. Coleman Memorial Laboratory
Saul and Ida Epstein Otoneurological Laboratory**

871 HSE

University of California at San Francisco

San Francisco, CA 94143

Telephone (415) 666-2511

April 29, 1985

Dr. Blake S. Wilson
Neuroscience Program Office
Research Triangle Institute
Research Triangle Park, NC 27709

Dear Blake,

This letter is written to acknowledge our continued enthusiasm for collaborating with you on your development of speech processors for cochlear prostheses, supported by the Neuroprosthesis Contract Program. We believe that progress in your initial contract period has been outstanding. Your construction of the Block Diagram Compiler speech processor model is of tremendous importance to our group. With the beautiful interface hardware that you have provided, we'll make powerful use of this instrument (with your collaboration) in the next UCSF experimental patient (to be implanted on May 8). Your progress in e-field modeling of intracochlear electrodes has also been outstanding, and highly relevant to your -- and our -- long range objective of efficiently defining what you have termed "stimulus primitives". We (especially Mark White) are making good use of these data already. I believe that with Mark White's psychophysical studies, a quantum leap has been achieved in the understanding of basic electrophysiological considerations underlying control of an intracochlear electrode array.

To further support the RTI-USCF collaboration, we are recruiting a speech psychophysicist to work with your speech processing model at UCSF. The search committee has now selected a candidate for this position. This young Ph. D., Dr. John Kingston, is thoroughly familiar with Eclipse systems and with ILS software, and should be of immediate help. If he accepts this position, he will join the group around June 1. With Mark White, a doctoral student (David Morledge) and our engineering personnel collaborating in these experiments, we believe that we can strongly support the evaluation of your model at UCSF, and will continue to use it heavily over your proposed renewal period.

As you know, we are also delighted to hear of your possible collaboration with the excellent research group at CID and the Otolaryngology Department at Washington University. This is another of the several research teams really capable of making substantial progress in prosthesis speech processor development. We shall support the Washington University group in whatever way we can from our end.

We are absolutely delighted to continue our collaboration with you, and, indeed, look forward to even stronger intergroup ties. Mark White's possible direct work with you on

consideration of electrode-nerve interface studies is a case in point. As you appreciate, Mark is the world's authority in this area re auditory nerve stimulation. This collaboration should be a highly fruitful one for all concerned.

Keep up the good work. We look forward to Charlie and you joining us in June for another intensive research period. Let's hope that this patient is one with good nerve survival!

Yours,



Michael M. Merzenich
Director, Coleman Laboratory
Professor, Physiology & Otolaryngol.

Duke University Medical Center

DURHAM NORTH CAROLINA 27710

DEPARTMENT OF SURGERY
DIVISION OF OTOLARYNGOLOGY
P. O. BOX 3805

OFFICE (919) 684-6968
APPOINTMENTS 684-3834

May 14, 1985

Mr. Blake S. Wilson, Director
Neuroscience Program Office
Research Triangle Institute
Research Triangle Park, North Carolina 27709

Dear Blake:

This letter is to indicate our enthusiasm for our continued collaboration with you on your projects for the development of speech processors for cochlear prostheses. We have been very impressed with the progress that you have made with your first contract for this project, and your work has been very influential in our efforts to begin a cochlear implant program at Duke University Medical Center. In response to your input to our program, Dr. David C. Sabiston, Jr., Chairman of the Department of Surgery, has authorized monies to assist in the construction of the interface hardware that you designed for testing patients with your Computer-based Simulator of Speech Processors. Your interface hardware and your computer simulation techniques will be essential components in our clinical implant program.

As you know, we are actively assessing applicants for a cochlear implant, and once a suitable candidate is implanted, we will use your computer-based methods to determine the signal processing schemes that are best for that patient. All the members of our team believe that your approach--that of assessing many signal processing schemes--will provide the most benefit to the cochlear implant patient.

During the course of assessing candidates for cochlear implants we shall make every effort to identify those patients who would be suitable experimental subjects, and we will actively seek funding for the implants of these patients. Such experimental patients will undergo tests described in your experimental protocol and in your contract applications to NIH as well as in other written material to us.

We think we have made a useful contribution to your project in the past, and we intend to contribute in the future. We have an active research program in the anatomy and physiology of the central auditory system. Members of this program are willing to assist you in any way they can. We would like to add that the long term goal of our research program is to attract additional scientists who would interface between our basic research program and the clinical application of cochlear implants.

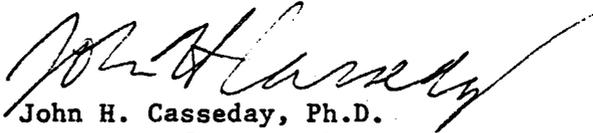
Mr. Wilson
Page 2
May 14, 1985

We look forward to the continuation of an extremely productive collaboration with you.

Sincerely,



William R. Hudson, M.D.
Professor and Chief



John H. Casseday, Ph.D.
Director of Research
Associate Medical Research Prof.

WRH/ts

Duke University Medical Center

DURHAM, NORTH CAROLINA
27710

DEPARTMENT OF SURGERY
CENTER FOR SPEECH AND
HEARING DISORDERS

TELEPHONE (919) 684-3859
P. O. BOX 3887

May 10, 1985

Blake S. Wilson, Head
Neuroscience Program Office
Research Triangle Institute
Research Triangle Park, NC 27709

Dear Blake,

I am writing to provide you with a formal statement of the enthusiastic support of the Speech Pathology and Audiology Programs for your development of speech processors for cochlear prostheses. We are delighted to have an opportunity to collaborate with you in this endeavor.

We believe that the close proximity of Duke Medical Center and RTI would allow us to work closely in providing a superior cochlear implant system for the profoundly hearing impaired. The Center for Speech and Hearing Disorders at Duke, as part of a major medical center, is providing ongoing clinical services to a large number of hearing impaired individuals. This should serve as an excellent source for cochlear implant candidates who would agree to serve as research subjects in your project. The newly established multi-disciplinary Center for the Severely Hearing Impaired at Duke should further enhance our ability to identify individuals who can profit from the UCSF - Storz cochlear implant device. We will make every effort to ensure that these patients are aware of the importance of your research and your need for research subjects.

I am personally pleased to be involved in your research efforts and will assume responsibility for coordinating the activities here at Duke. I will also provide assistance in the development of testing strategies and protocols.

The entire staff here at the Center for Speech and Hearing Disorders will cooperate with you in the conduct of your research activities. As you are aware, we have allocated space for the testing of implant candidates and have arranged for renovations to provide suitable housing for the Eclipse computer. I believe that Duke financial assistance in the duplication of the RTI computer here in the Center also attests to strong support for your project.

Blake S. Wilson
May 10, 1985

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We are most pleased that there is an opportunity for us to become more closely involved with your research endeavors. We strongly believe that these efforts will result in a speech processor which will be of great benefit to the profoundly hearing impaired. We are delighted to work with you toward this most important goal.

Sincerely,



Bruce A. Weber, Ph.D.
Associate Professor

BAW:mf

V. Statement of Work, Schedule and Budget

Independently and not as an agent of the Government, Research Triangle Institute (RTI) will exert its best efforts to design and develop speech processors for use with auditory prostheses. The Description of Work as given in RFP No. NIH-NINCDS-85-09 is acceptable to RTI, and is reproduced below for reference.

Specifically, RTI will:

- A. Design and develop a computer-based, multichannel waveform generator which when coupled with the collaborating investigators' multichannel neural stimulators will permit studies on:
 - 1. Improved temporal and spatial resolution of stimuli.
 - 2. The extension of the dynamic range of intensity coding.
 - 3. The extension of the range of frequency coding.
 - 4. Methods of preserving the subjective sensation of constant pitch while changing the subjective sensation of loudness.
 - 5. The factors which control the subjective sensation of pitch.

- B. Design and develop a computer-based, multichannel auditory signal processor for use in evaluating promising speech extraction and stimulus encoding schemes.
 - 1. The processor shall contain microphones to capture speech signals under various environmental conditions and convert them to electrical signals for preprocessing.
 - 2. Sufficient preprocessing circuitry shall be included to compensate for known psychophysical limitations of present multichannel auditory prostheses such as limited dynamic range of input

signals as determined by subjective loudness and limited rate pitch discrimination.

3. Devise hypotheses of potentially feasible speech processing schemes based on presently known psychophysical data from auditory prosthesis implant patients and develop software for testing them using a laboratory-based computer such that:
 - a. Their key elements can be varied independently.
 - b. They can be evaluated in human subjects in conjunction with current designs of single electrode or multiple electrode auditory prostheses.
 - c. They can be used to further evaluate multielectrode psychophysical characteristics including complex interelectrode interactions.
 - d. Comparisons can be made between different speech processing schemes in the same implant patient.
 - e. The essence of each of the speech processing schemes could be reduced to a real time, hardware based, wearable speech processor.

- C. Design and fabricate wearable speech processors based on the results obtained with the computer-based simulated designs such that:
 1. They are designed for specific patients with single or multielectrode auditory prostheses.
 2. They are human engineered with respect to weight, durability, and panel component selection and placement.
 3. They take advantage of the implanted electrode configurations.
 4. They operate in real time.
 5. They can be used for studying the long-term effects of learning.

D. Supply at least two of these wearable speech processors to the Project Officer by at least three years after the start of the contract.

1. Include details of suggested procedures for evaluating these speech processors.

E. Assist the collaborating human subject evaluation team in implementing the above mentioned waveform generator, computer-based signal processor and the wearable speech processors.

Our general plan for the first year of work in the "present scope" contract includes the following:

1. Continue work on software to generate the stimuli for studies on "stimulus primitives";
2. Complete work on the block-diagram compiler, to improve the user interface and to incorporate the additional modules required for simulation of all processing strategies outlined in sections III.B and III.C;
3. Automate, to the extent possible with funds available for this "present scope" contract, procedures to obtain measures of psychophysical performance and speech understanding;
4. Collaborate, to the extent possible with funds available for this "present scope" contract, in the conduct of initial tests at UCSF and DUMC, to assist the teams at these centers in the evaluation of stimulus primitives and speech processing strategies;
5. Continue development of 80C31-based processors,

including implementation of processors that have dual 80C31s or one 80C31 and a multiplier chip;

6. Define design changes that would be required to implement strategies outlined in section III.C of this proposal using our 80C88-based processors; and
7. Prepare and submit reports of progress at the end of each quarter, as specified in the RFP.

Work in years 2 and 3 would be mainly directed at completion of tasks 1, 3, 4, and 5 above, and at design and construction of hardware processors based on the most promising results obtained in the evaluation studies by the collaborating psychophysical teams.

The budget requested for this contract is slightly greater than the budget for our present contract because our experience with the present contract indicates that (1) more frequent travel to San Francisco will be required to maintain a highly productive collaboration with the UCSF team and (2) we need somewhat greater support for development of hardware while maintaining our commitment to define the classes and parameters of processor design that will allow full recognition of speech for recipients of multichannel auditory prostheses.