DEVELOPMENT OF A SPEECH PROCESSOR FOR LABORATORY EXPERIMENTS WITH COCHLEAR IMPLANT PATIENTS

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ABSTRACT

A laboratory speech processor based on Motorola's DSP56002 chip was developed for cochlear implant research. The hardware of the speech processor is described in detail and the safety issues associated with electrical stimulation of the auditory nerve are discussed. The speech processor is capable of providing high-rate stimulation to six electrodes using short biphasic pulses presented either simultaneously or in an interleaved fashion. Different speech processing algorithms including the Continuous Interleaved Sampling (CIS) strategy were implemented in this processor and tested successfully with cochlear implant patients.

1. INTRODUCTION

Cochlear prostheses are devices that provide partial hearing to deaf people through electrical stimulation of the auditory nerve (Wilson *et al*, 1988). A typical device consists of a speech processor, a behind-the-ear unit and a cochlear implant (Tierney *et al*, 1994). Sound is picked up by a microphone in the behind-the-ear unit and sent to the speech processor. Depending on the speech processing algorithm being implemented, the speech processor extracts various parameters from the input signal and determines the amplitude of the current pulses to be sent to the implant (Loizou 1998). The amplitude information is transmitted to a receiver/stimulator circuit implanted under the scalp of the patient. This circuit sends current pulses to the electrodes implanted in the cochlea of the patient. As a result, the auditory nerve is excited and transmits nerve pulses to the brainstem and brain, where they are interpreted as sound (Ay *et al*, 1997).

Speech processors for cochlear implants can be divided into two groups, wearable processors and laboratory processors. Different factors are important in the design of each type. Wearable speech processors are used daily by cochlear implant patients. Therefore, size and weight are important design parameters. Most patients wear their processors concealed in their clothing, which makes it difficult for them to change batteries in social environments. Therefore, in order to increase battery life, wearable speech processors are designed to consume little power. Because of these limitations, wearable speech processors are optimized to implement a specific signal processing technique. Only a limited number of parameters associated with this technique can be modified. In the case of a laboratory speech processor, however, much more versatility can be incorporated in the design, because size, weight and power consumption are not significant design parameters.

The laboratory speech processor described here is based on the design of a wearable speech processor (François *et al*, 1994; Boëx *et al*, 1996). It was developed from the original design through a number of modifications and additions. Section 2 describes the electronic components of the laboratory speech processor, with emphasis on these changes. As with all medical instruments, the most important issue in the design of a laboratory speech processor for cochlear implants is the safety of the patient. AC currents with very low amplitudes flowing through a human body may result in permanent injury or death. Therefore, it is essential to take all precautions to ensure the safety of the patient. The safety features of the laboratory speech processor are described in Section 3.

2. SPEECH PROCESSOR HARDWARE

The electronic components of the laboratory speech processor are shown in Fig. 1. The circuits, which receive the audio signal, process it and generate electrical stimuli for cochlear implants are discussed in this section. The circuits related to safety issues are discussed in the next section.

The laboratory speech processor is based on Motorola's DSP56002 digital signal-processing (DSP) chip. The circuit board in the center of Fig. 1 contains the DSP chip and most of the digital circuitry. The random-access memory (RAM) chips are used to store large look-up tables, which are not frequently accessed by the DSP chip. The internal memory of the chip is reserved for more frequently accessed data, such as coefficients of digital filters. The circuit board on the left-hand side of Fig. 1 contains the analog input circuit and the one on the right-hand side contains the output circuit. The full-duplex, synchronous, serial interface of the DSP chip is used to receive data from the input circuit and to transmit data to the output circuit. All other communication between the DSP chip and the rest of the speech processor is established over general-purpose input/output lines.

The analog input circuit (Fig. 2) consists of an audio multiplexer, several fixed-gain amplifiers, one variable-gain amplifier, three anti-aliasing filters and an analog-to-digital converter (ADC). The audio multiplexer is used to select the source of the audio input signal to the speech processor. A toggle switch on the control panel (Fig. 3) is used to select a built-in microphone or some other device connected to the auxiliary input jack of the speech processor as the source of the input signal. It is also possible to mix the signals from the microphone and the auxiliary device to generate the input signal. The

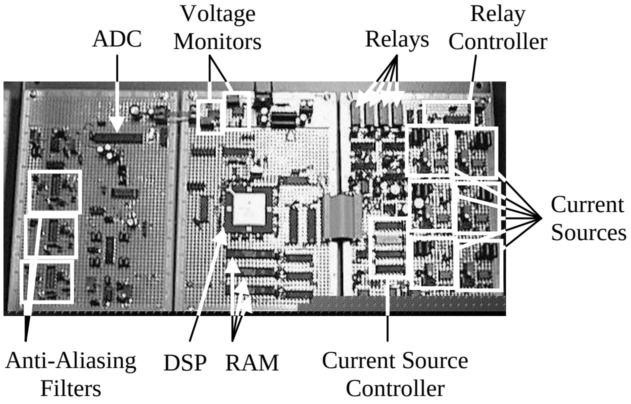

Figure 1. Electronic components of the speech processor.

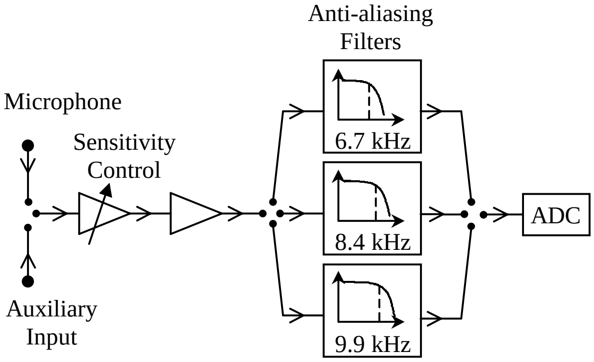

Figure 2. Schematic diagram of the input circuit.

variable-gain amplifier is used to match the range of the input signal to the range of the ADC. The gain of this amplifier is adjusted with the sensitivity knob on the control panel (Fig. 3).

The wearable speech processor, from which the laboratory processor was developed, was designed to operate with a fixed bandwidth of 6.7 kHz. In order to be able to study the effects of bandwidth on the performance of cochlear implant patients, two additional anti-aliasing filters were designed with cutoff frequencies of 8.4 kHz and 9.9 kHz. The analog input signal is fed through one of these anti-aliasing filters, according to the position of the bandwidth switch on the control panel (Fig. 3). This signal is sampled and digitized by the DAC. The sampling rate is controlled by the DSP chip. A sampling rate of 22 kHz is used for all bandwidth settings. Once the signal is digitized, it is transmitted to the DSP chip, where it is processed according to the particular signal-processing technique being implemented.

The laboratory speech processor is also capable of receiving digital data files from a Personal Computer (PC). The Motorola standard On-Chip-Emulator (OnCE) port of the DSP chip is connected to a Motorola command converter board, which, in turn, is connected to a PC. Digitized speech samples can be downloaded via this link to the RAM in the speech processor or they can be preprocessed in the PC and the resulting output representing the amplitude of the current pulses to be sent to the electrodes can be downloaded. A six-bit digital port connected on one end to the general purpose I/O pins of the DSP chip and on the other end to a digital I/O card installed in the PC is used to send file information to the speech processor. Each state of this port corresponds to a different data file stored in the PC. Thus, the speech processor can read up to 63 different data files from the PC (One state is used to control program flow.).

The wearable speech processor, from which the laboratory processor was developed, was designed for the Ineraid cochlear implant. This implant does not have a receiver circuit and the current pulses have to be generated in the speech processor and sent directly to the electrodes. The wearable speech processor was optimized to implement the CIS strategy, where the electrodes are stimulated one at a time. Therefore, it has only one current source. In the laboratory speech processor, six current sources were used, in order to be able to implement other strategies, where two or more electrodes are stimulated simultaneously. Each of these current sources is built around a digital-to-analog converter (DAC). An interface circuit was designed to control the flow of information from the DSP chip to the DACs (Fig. 4). During program execution, the DSP chip pulls high one of the address lines, A0 to A5, to select the corresponding DAC. Then, it transmits a data word via the serial data line. The data word is sent to all the DACs, but only the one that is selected receives the serial clock and shifts the data word in; the others ignore it. Then, the DSP chip sends out the latch signal, which is received only by the selected DAC. Upon the reception of the latch signal, the DAC starts the conversion of the data word. This procedure results in the update of the output of a single current source only. It is necessary to update the outputs of multiple current sources simultaneously to implement some variations of the CIS strategy. To do that, the DSP chip selects the first DAC to be updated and sends to it a data word, but no latch signal. Then, it selects the second DAC to be updated and sends to it a different data word. Finally, it selects both DACs and sends out the latch signal. Both DACs receive the latch signal, start converting their respective data words and their outputs are updated simultaneously. The difference in the conversion time of DACs can be ignored, because it is very small compared to the time frame of the electrical stimuli generated for cochlear implants.

3. SAFETY OF THE PATIENT

International safety regulations state that a medical device, which is semi-permanently connected to a human subject, may not have a direct connection to ground (Prutchi, 1996). Therefore, a battery is utilized to power the laboratory speech processor. The PC that is part of the experimental setup is powered through the electrical wiring of the laboratory. The patient is parasitically coupled to various sources of noise in the laboratory, such as the fluorescence lamps and the PC monitor. This configuration, together with the possibility of the generation of static charge through friction, raises the issues of leakage currents, electrical isolation, shielding and grounding. In the following sections, these issues are addressed and the safety features of the laboratory speech processor are described.

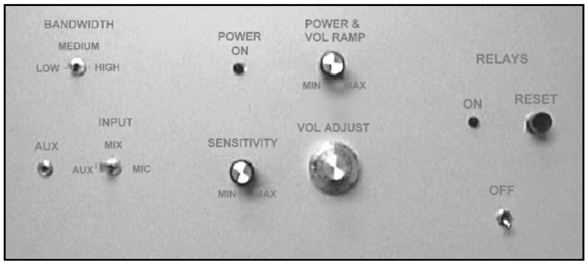

Figure 3. Control panel of the speech processor.

A. Electrical Isolation

It is dangerous to make an electrical connection between a patient and an instrument that is powered through the electrical wiring of a building. In case of a failure in the instrument, the patient may be exposed to deadly voltages. Any parasitic voltage appearing on the body of the patient may cause a current leakage from the body, through the instrument, to ground. Therefore, the patient must be electrically isolated from all instruments connected to ground. The speech samples used to test the speech recognition ability of cochlear implant patients are stored in digital format. It is possible to play these samples through a loudspeaker and use the built-in microphone of the speech processor to generate an analog input signal. However, in the absence of a laboratory with appropriate acoustic properties, the samples have to be sent from the PC directly to the speech processor in form of electrical signals. These signals can be analog or digital, and a different electrical isolator must be used in each case. For analog signals, a wide-band, unity-gain isolation amplifier is used to provide electrical isolation. This amplifier is based on an OP-AMP (Burr Brown ISO100), in which the input and the output sections are isolated from each other. The isolation is achieved by the coupling of electrical signals through light, using light-emitting diodes and photodiodes. The OP-AMP is powered by the same battery that powers the speech processor. The battery voltage is magnetically coupled to the supply voltages of the input and output sections of the OP-AMP through a dual-isolated DC/DC converter (Burr Brown 722). Digital signals between the PC and the speech processor are fed through isolated digital couplers (Burr Brown ISO 150), which provide electrical isolation between the two devices.

B. Shielding

Electrical isolation protects the patient from being exposed to high voltages and prevents leakage currents from reaching dangerous levels. However, some current will leak through any isolation. In some studies, it is necessary to further reduce leakage currents. For example, in the measurement of evoked potentials, leakage currents may contaminate the data read from the electrodes on the patient's body. In the case of cochlear implants, the path of any current leaking through the patient's body will include the auditory nerve. A current flowing through the auditory nerve will be perceived by the patient as a sound and the results of speech recognition studies will be affected.

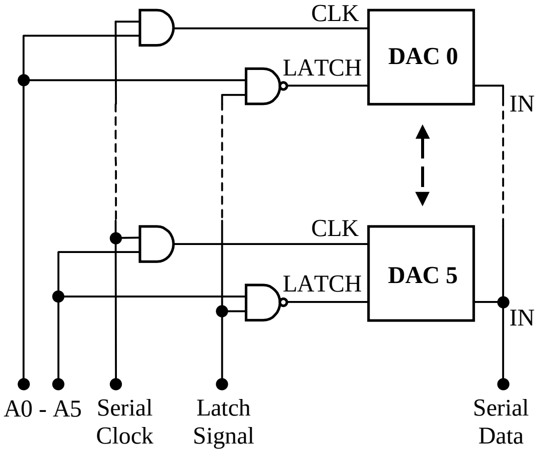

Figure 4. Current-source control circuit.

The main source of leakage currents are the parasitic voltages appearing on the body of the patient and on the electrical circuits, which act as an antenna when exposed to the electromagnetic radiation in the laboratory environment. The laboratory speech processor was enclosed in a metal instrument case and the case was connected to ground. The battery that powers the processor was wrapped in aluminum foil and the foil was connected to ground. With this configuration and the electrical isolation, the current leakage from the speech processor to ground was reduced to below 0.1 μA.

C. Grounding of the Patient

Since the body of the patient can not be shielded, it will pick up some environmental noise, unless the experiments are conducted in a shielded laboratory. Therefore, it is desirable to provide an alternative path to ground, so that current will not leak through the auditory nerve, the processor and the isolation. This path is provided by a floor mat and a desk pad (Fig. 5). The mat and the pad have conductive surfaces and they are connected to ground through one-megaohm resistors. The mat is placed under the patient's chair and the pad is placed on the desk in front of the patient. The patient is instructed to keep at least one wrist on the desk pad at all times. The paths through the mat and the pad also allow any static charge, which may accumulate on the body of the patient, to flow to ground in a controlled way.

D. Relay Disconnect Circuit

The outputs of the current sources and the current return path are connected to the cochlear implant of the patient through a set of relays (Fig. 1). In case of an emergency, the experimenter can open the relays with a toggle switch on the control panel of the speech processor (Fig. 3). The patient can do the same by pressing a pushbutton switch on a small box placed directly under the hand of the patient that is resting on the desk pad (Fig. 5). The relay circuit is designed so that, once the relays are opened, they will remain open, regardless of the position of the switches, until the reset button on the control panel is pushed. The relay circuit is powered by a separate battery pack (Fig. 5)

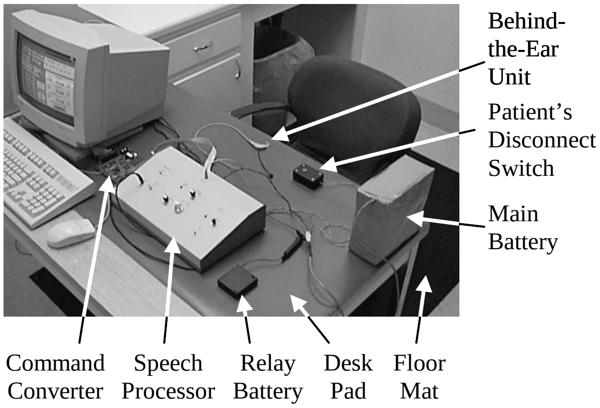

Figure 5. Experimental setup.

and it will continue to operate even if the main battery and the speech processor have failed. If the voltage of this battery pack drops, or if it is disconnected, the relays will automatically open (normally-open configuration).

E. Voltage Monitors

The voltage of the main battery that powers the laboratory speech processor is monitored by two circuits (Fig. 1). These circuits compare the battery voltage to two preset threshold levels. If the battery voltage drops below one of these threshold levels, the corresponding voltage monitor circuit changes the state of its output. The outputs of the monitor circuits are connected to two general-purpose digital input/output pins of the DSP chip. During program execution, the DSP chip routinely checks these pins. The first monitor circuit changes its output when the battery voltage drops to within 0.5 V of the minimum voltage required for the stable operation of the processor. If the DSP chip detects a change in the output of this circuit, it causes the power indicator on the control panel (Fig. 3) to start to blink and it gradually reduces the amplitude of the current pulses sent to the electrodes in the cochlear implant. The second monitor circuit changes its output when the battery voltage drops below to the minimum voltage required for the stable operation of the speech processor. If the DSP chip detects a change in the output of this circuit, it disconnects the current sources from the electrodes and turns the speech processor off.

4. RESULTS

The speech processor described here was successfully used in laboratory experiments with six patients implanted with the Ineraid cochlear implant. The effect of several parameters of the CIS speech processing strategy on the speech recognition performance of the patients has been investigated (Loizou and Poroy, 1999). These parameters include the rate and the width of the current pulses sent to the electrodes, the shape of compression functions, the order of digital filters, the resolution of the output and the order in which the electrodes are stimulated. Currently the speech processor is being used to develop new speech processing algorithms for cochlear implants.

REFERENCES

Ay, S., Zeng, F. -G. and Sheu, B. (**1997**). "Hearing with bionic ears: Speech processing strategies for cochlear implant devices," IEEE Circuits & Devices Magazine, May 1997, 18-23.

Boëx, C., Pelizzone, M. and Montandon, P. (**1996**). "Speech recognition with a CIS strategy for the Ineraid multichannel cochlear implant," Am. J. Otol. **17**, 61-68.

François, J., Tinembart, J., Bessat, C., Leone, P., Rossman, F. and Pelizzone, M. (**1994**). "Implants cochle aires: Un processeur portable pour le développement de l'algorithme CIS," in Actes de le conférence DSP 94, Paris

Loizou, P. (**1998**). "Mimicking the human ear: An overview of signal-processing strategies for converting sound into electrical signals in cochlear implants," IEEE Signal Processing Magazine **5**, 101-130.

Loizou, P. and Poroy, O. (**1999**). "A parametric study of the CIS strategy," presented at the 1999 Conference on Implantable Auditory Prostheses, Asilomar, Monterey, CA.

Prutchi, D. (**1996**). "Designing medical electronic-device prototypes," Circuit Cellar Ink **74**, 26-40.

Tierney, J., Zissman, M. and Eddington, D. (**1994**). "Digital signal processing applications in cochlear-implant research," The Lincoln Laboratory Journal **1**, 31-62.

Wilson, B., Finley, C., Lawson, D. and Wolford, R. (**1988**). "Speech processors for cochlear prostheses," Proc. IEEE **9**,1143-1154.