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Application of Digital Signal Processor for the Nucleus 22 Channel Cochlear Implant System

Shigeyoshi KITAZAWA*, Mitsuaki NAITO**, Koushi YAMADA***
and Shuji DOSHITA****

ABSTRACT

Nucleus 22 channel cochlear implant system extracts features with an analog electric circuit. We replaced analog with digital processing and devised an acoustic simulator to evaluate the system. Our system consists of three parts, a DSP (Digital Signal Processor) board, a BCG (Burst Code Generator) and an acoustic simulator. The DSP board is not only a replacement of the analog circuit with a digital signal processor TMS32010, but provides also many other possibilities of advanced processing algorithms. The BCG was realized a fully compatible interface with the conventional implant system, so the implanted receiver-electrode units can be arbitrarily controlled from the DSP. The acoustic simulator represents the psychological effects for the subject wearing the implant system which excites the characteristic frequency resonator by the stimulus pulse for each channel. The design of our system is described in this paper.

1. INTRODUCTION

Cochlear implants are in common use for patients who cannot hear anything using conventional hearing aids. Cochlear implants create a sensation of sound by direct electrical stimulation of the auditory nerve to help speech understanding. Nucleus 22 channel cochlear implant system consists of a WSP (Wearable Speech Processor) and a RSU(receiver stimulator unit) with electrodes. The WSP extracts features of speech sound, such as formants and pitch, and transmits it to the RSU implanted inside the body. The transmitted data encodes selection of the electrode, intensity of current and duration. A selected electrode drives the auditory nerve by an electrical stimulation. Original WSP use analog electric circuits for extracting features.

* Shigeyoshi KITAZAWA (北澤茂良): Associate professor, Department of Computer Science, Faculty of Engineering, Shizuoka University
** Mitsuaki NAITO (内藤光明): Graduate student, Department of Computer Science, Faculty of Engineering, Shizuoka University
*** Kousi YAMADA (山田耕史): Engineer, Department of Computer Science, Faculty of Engineering, Shizuoka University
**** Shuji DOSHITA (堂下修司): Professor, Department of Information Science, Faculty of Engineering, Kyoto University

We replaced the WSP by a digital signal processor in order to improve quality and accuracy for listening. We devised an evaluator system which consists of a DSP board, a BCG(Burst Code Generator) and an acoustic simulator. The DSP board processes the input speech and extracts acoustic features by numerical algorithms. The BCG generates coded burst pulses which are transmitted to the RSU. The acoustic simulator represents the psychological effects by creating sound. Description as follows.

2. COMPONENTS

As a target we used a Nucleus 22 channel cochlear implant by Cochlear corporation. A conventional DSP(Digital Signal Processor) TMS32010 composes a major part of a speech processor. A system overview is shonw in Fig. 1. Our system consists of three parts, a DSP board, a BCG(Burst Code Generator) and an acoustic simulator. The original transmitter coil of the implant system was used for coupling with the RSU(receiver stimulator unit). The original WSP was separated into a DSP board and a BCG to change the method of processing. The hardware components of the system are described as follows.

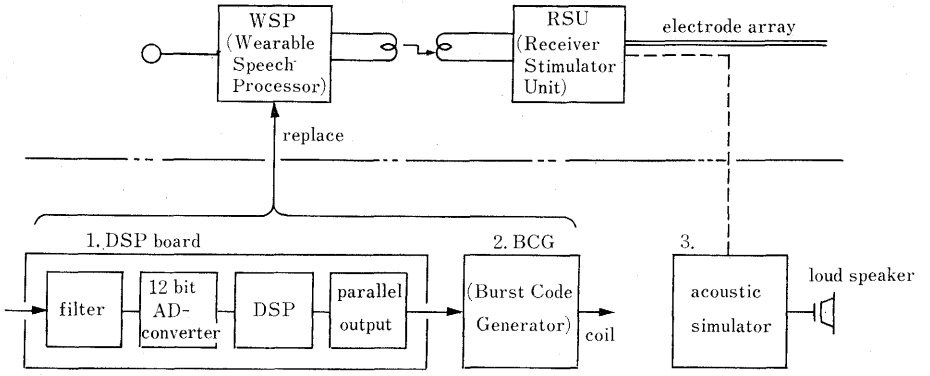


Fig. 1. System overview. The upper illustration shows an original cochlear implant system which consists of WSP (Wearable Speech Processor) and RSU (Receiver Stimulater Unit) with electrodes. The lower illustration shows our system which consists of three parts; a DSP board, a BCG (Burst Code Generator) and an acoustic simulator. The original WSP is functionally equivalent to the DSP board plus BCG.

2.1 *Speech processing part (a DSP board)*

This part processes the input speech and extracts acoustic features. A first generation digital signal processor TMS32010 is a speech processor on the module. The evaluation board used for this preliminary study consists of a DSP(clocked 20 MHz), program memory(RAM), two low-pass filters, a 12 bit AD converter, a 12 bit DA converter, parallel inputs and parallel outputs. The sampling rate was set to 10 kHz. Certain stages of the processing was monitored by a DA converter.

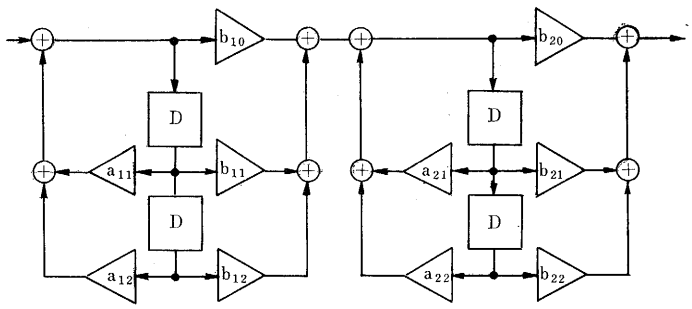
Table 1. Coefficients of filters illustrated in Fig. 2.

(a) coefficients of 270 Hz LPF of the oder 4.

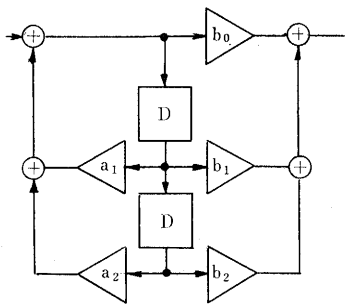
	270 Hz LPF		
a_{11}	1.884215	a_{21}	1.926289
a_{12}	−0.891842	a_{22}	−0.953963
b_{10}	0.001907	b_{20}	0.006919
b_{11}	0.003814	b_{21}	0.013837
b_{12}	0.001907	b_{22}	0.006919

(b) coefficients of 1.5 kHz HPF, 850 Hz LPF and 480 Hz HPF of the oder 2.

	1.5 kHz HPF	850 Hz LPF	480 Hz HPF
a_1	0.877352	1.326983	1.670414
a_2	−0.417811	−0.565661	−0.741874
b_0	0.573791	0.059669	0.853072
b_1	−1.147582	0.119339	−1.706144
b_2	0.573791	0.059669	0.853072



IIR filter of order 4



IIR filter of order-2

Fig. 2. Block diagram of digital filters.

Processed data and commands were sent to the BCG via 32 bit parallel outputs.

DSP programs are written in assembly language. Programs are edited on the personal computer and transmitted to the DSP board via serial RS232C data communication port (9600 bps). We use an algorithm which is similar to the original WSP [1]. F1(first formant) and F2(second formant), were extracted and F0(pitch) was estimated. Frequencies were estimated by zero crossings of the filtered signal: 1.5 kHz HPF(high pass filter) for F2, 850 Hz LPF(low pass filter) and 480 Hz HPF for F1 and 270 Hz LPF for F0. The amplitude for F2 and F1 were estimated by the peek hold of each signal after filtering. Table 1. shows coefficients of filters illustrated in Fig. 2. Each filter is a Chebyshev IIR filter of order 4 for 270 Hz LPF and order 2 for others.

2.2 *BCG(Burst code generator)*

This part generates burst codes fully compatible with the Nucleus implant system and sends them to the transmitter coil[2]. The BCG consists of an oscillator, RAM(2 k x 16bit), a programmable counter, a readout sequence counter and an address counter. Pulse on-off instructions and pulse counts are written in the order into RAM. When the start command is given to the BCG to trigger an auto-readout sequence, the instructions are read out in order from the RAM to the programmable counter to generate the exact number of pulses. The block diagram is shown in Fig. 3. For each electrical stimulation, the BCG creates a formatted packet with a carrier of 2.5 MHz. The format of the data transmission packet is shown in Fig. 4. One frame includes 6 bursts. The first burst, which includes 4 pulses, is synchronization burst that shows the start of the frame. The second burst specifies the main electrode number. The third burst is the mode burst that specifies the difference

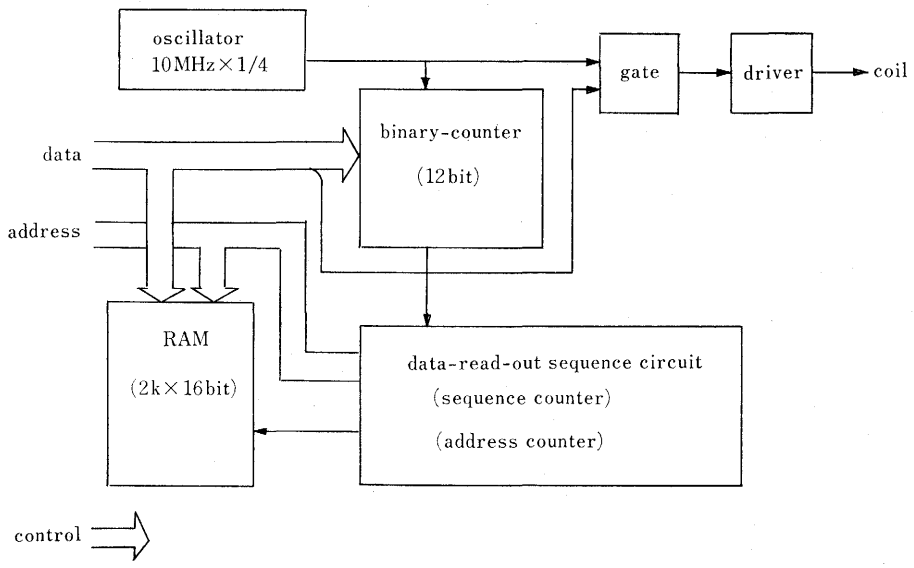


Fig. 3. Block diagram of the BCG.

between the first electrode and the return electrode. The fourth burst indicates stimulus current to the implant. The 5th and 6th bursts specify the durations of the leading pulse and the second pulse respectively.

Fig. 5 shows the data format of the BCG. The lower 12 bits [bit 0–11] indicate number of pulses to be generated. Bit 14 is assigned to the pulse on-off flag. If this bit is set to '1', it allows the specified number of 2.5 MHz pulses to pass to the transmitter coil else it inhibits the passing. Bit 15 is assigned to the end flag. If this bit is set to '1', the auto-read-out sequence is stopped, else the address counter is incremented to continue at the next address. Fig. 6 shows an example of the stored

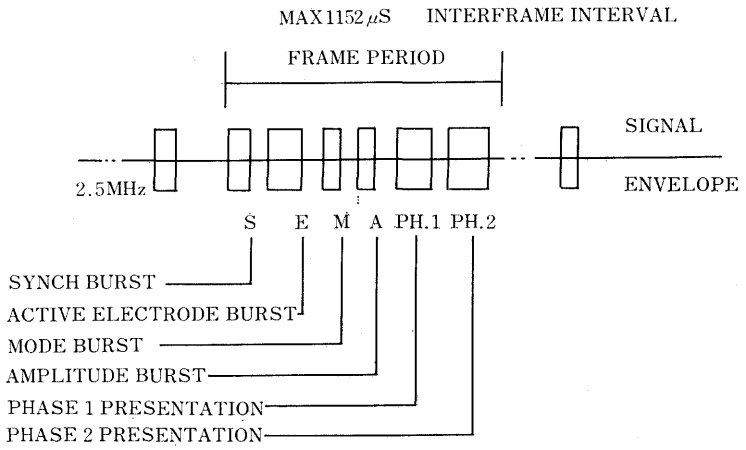


Fig. 4. Format of data transmission packet.

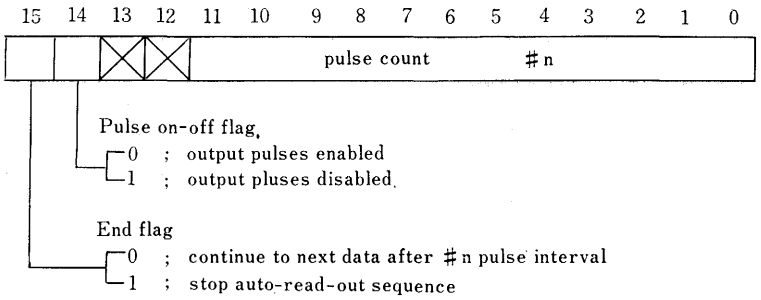


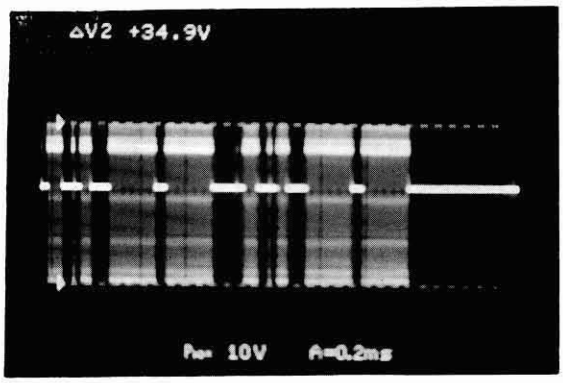
Fig. 5. Data format of the BCG.

Memory Address	Data
0	4004H
1	0028H
2	40A5H
3	0070H
4	401EH
11	8000H

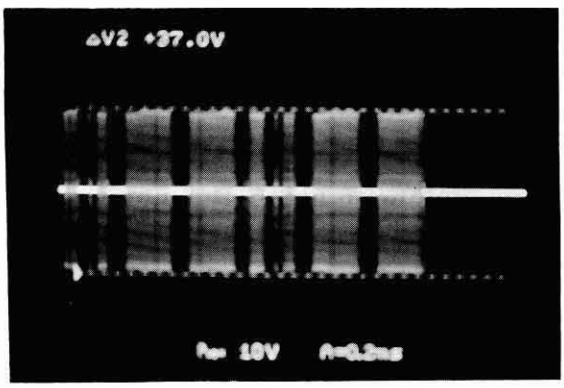
Fig. 6. Example of stored data in the BCG memory.

data in the BCG memory. The item at address 0 specifies a start burst with 4 pulses and pulse-on sign. The second item at address 1 specifies 40 pulse counts of silent interval between the first burst and the second burst. The third item at address 2 specifies the acitve electrode number 20 with 165 pulses, and so on. The last item at address 11 specifies the end of data, by the end-flag '1', which terminates the auto-read-out sequence. The BCG stops until the next trigger is received.

Fig. 7 shows pictures of the burst signals from the speech processor measured at the transmitter coil. The upper picture is taken from a genuine analog type speech processor. The lower picture is taken from our system. Fig. 8 shows electro-current signal measured at the electrode of a reciever stimulator(RSU). The upper picture is taken from a genuine analog type speech processor. The lower picture is taken through this system. These figures assure electric and logical compatibily between two systems.

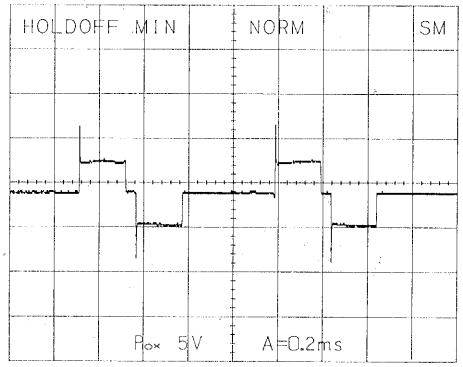


(a) ANALOG PROCESS

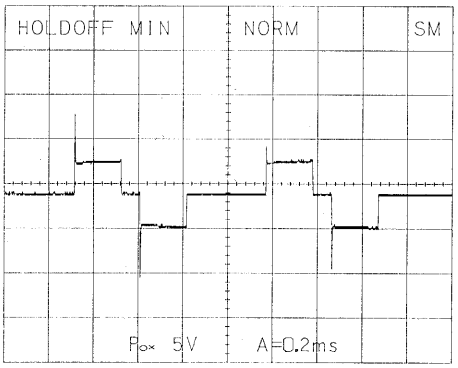


(b) DIGITAL PROCESS

Fig. 7. Burst signals at the transmitter coil are displayed on an oscilloscope screen. Two packet of burst signals are shown on each picture. The first packet carries F2(second formant) information. The second packet carries F1(first formant) information. The upper picture is taken from a genuine analog type speech processor (WSP). The lower picture is taken from our system.



(a) stimulate pulse (ANALOG PROCESS)



(b) stimulate pulse (DIGITAL PROCESS)

Fig. 8. Electro-current signal at the electrode of a receiver stimulator unit(RSU) with 10 k ohm resistor for a dummy load is plotted with a digital oscilloscope HITACHI VC-6045. The upper picture is the current with a genuine analog type speech processor, and the lower with this system. These figures assure electric and logical compatibily between two systems.

2.3 *Acoustic simulator*

The acoustic simulator represents the psychological effects for the subject wearing the implant system. In the previous report[3], we have shown the possibilities of acoustic simulation in a form of vocoder model. This time we took more realistic approach using a genuine implanted electrode. The stimulus pulse for each channel excites the characteristic frequency resonator.

Fig. 9 shows a picutre of the transmitter coil and the receiver with attached probes. A stimulate pulse applied to the electrode produces a short tone of the frequency characteristic to the electrode position along the cochlea. How far this will aid research in cochlear implants has yet to be seen in further evaluation of phonological information. The simulator definitely produces different effects if we change parameters or algorithms of our DSP system.

The block diagram is shown in Fig. 10. Electric current flow between different electrodes is initiated as a biphasic pulse. Each electrode of the RSU is connected

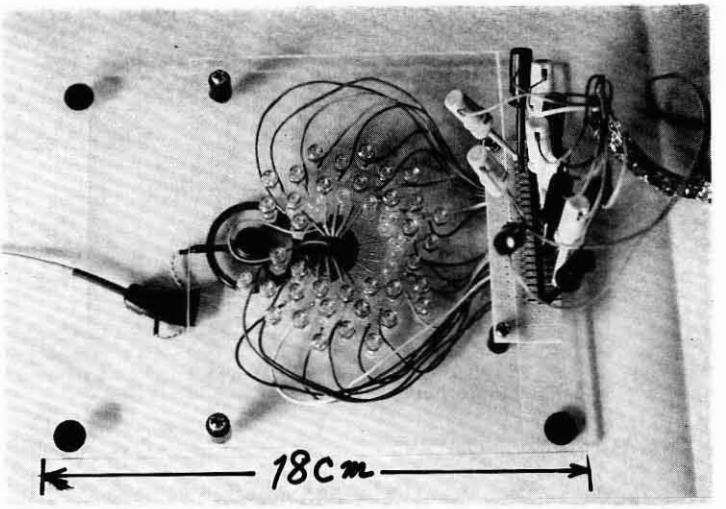


Fig. 9. Transmitter coil and the receiver chip with attached probes.

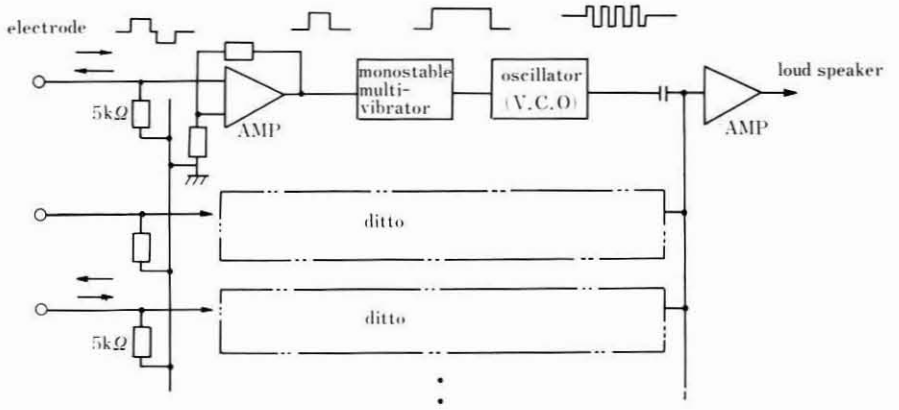


Fig. 10. Block diagram of the acoustic simulator.

to the simulator's common ground by a 5 k ohm resistor in order to terminate at 10 k ohm, an approximate human body electric resistance, between a pair of electrodes. The electric voltage on the resistor is fed to an operational amplifier. If the input voltage is greater than the threshold level(about 100 mV for 20 micro-ampere), the amplifier sends a short pulse(about 100 micro-seconds), with the same duration as one phase of a biphasic pulse. The short pulse is then extended to a 20 ms pulse by a monostable multi-vibrator. From this pulse, an oscillator connected to each multi-vibrator generates 20 ms tone. These tones are summed and amplified to a loud-speaker via an audio amplifier.

3. CONCLUSIONS

There does not seem to be any serious problem in using digital signal processing for speech processor. Our experiences are as follows:

1) Distortion of signal. In the case of an analog filter, the major source of signal distortion is the saturation of the amplifiers. In digital processing, distortion of the output signal appears in case of numerical over flow. The signal deteriorates suddenly.
2) Easiness of tuning. In digital signal processing, most of the tuning data is stored numerically. Numerical setting is easier than critical analog tuning.
3) Flexibility. Digital signal processing brings a lot of flexibility. We can install another processing by changing only the algorithm and using the same hardware.

DSP is very suitable for sound signal processing by using calculation. However conventional WSP have many additional functions which is not accomplished current DSP board, not only extracting features of sound but also referencing patient-specific loudness maps or others.[4] In future, associating by the DSP with other auxiliaries, like general processor, seems promising.

4. Acknowledgement

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