variation in the final impedances across the array, and one unusually high value had clinical significance.

REFERENCES


MANIKIN AND COCHLEAR IMPLANT PATIENT TEST RESULTS WITH A PORTABLE ADAPTIVE BEAMFORMING PROCESSOR TO SUPPRESS THE EFFECTS OF NOISE

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A simple adaptive beamformer (ABF) was implemented in a real-time portable speech processor and tested with four cochlear implant patients. The ABF algorithm used signals from only two microphones — one behind each ear — to attenuate sounds not arriving from the direction directly in front of the patient, and was compared with a strategy in which the two microphone signals were simply added together (two-microphone broadside strategy). Tests with the four patients were conducted in quiet and in noise. Results at a 0 dB signal-to-noise ratio showed large improvements in speech intelligibility for all patients, when compared to the two-microphone broadside strategy. Physical measurement of the directional characteristics of the ABF processor were made with a Kemar manikin. The effects of reverberation were explored by placing the manikin in different acoustic environments and observing the attenuation of the noise slope at various angles. A near-anechoic environment allowed the noise to be attenuated by as much as 21 dB, whereas in a highly reverberant concrete stairwell, the ABF processor was unable to provide any directional gain beyond about 3 dB.

INTRODUCTION

The presence of moderate to high levels of background noise continues to pose problems in speech perception for most cochlear implant patients. Very few cochlear implant patients have useful levels of open-set speech understanding when interfering noise is similar in level to the speech signal of interest. Multiple microphone techniques are being developed in the hope of reducing some of the effects of noise. These are generally divided into fixed and adaptive methods. Of these two, the former allows simpler design and consequently modest power consumption. The adaptive techniques are generally computationally intensive and do not lend themselves to real-time portable applications so readily. They can, however, offer performance advantages as far as noise suppression is concerned, and as technologies evolve, we may expect to see more real-time adaptive applications, even in power-sensitive devices such as hearing aids. We present here one such adaptive noise reduction scheme and its implementation in a portable digital signal processor for use with cochlear implant patients. More detailed results have been reported in van Hoesel and Clark.

ADAPTIVE BEAMFORMING PROCESSOR

The noise reduction technique called adaptive beamforming (ABF) usually refers to methods that make use of the differences between the signals at multiple microphones to preserve or attenuate signals from particular directions. In our application, the direction for which we want to preserve the signal is held fixed directly in front of the patient. The processor attempts to attenuate any sound from other directions. In implementing real-time ABF for cochlear implant patients, we have chosen to test a two-microphone array with one microphone worn behind each ear. This avoids the problem of mounting more than two microphones for patients not wearing glasses, and reduces power consumption to a level manageable for a portable device. Evaluation of the processor has been divided into tests with a Kemar manikin and tests with implant patients. It is well known that the performance of ABF deteriorates with reverberation. Accordingly, a Kemar manikin was fitted with the ABF processor and performance measurements were made in a number of acoustic environments. Four cochlear implant patients were then tested with the processor under living room conditions. Tests were performed in quiet as well as in noise to ensure that in quiet, no degradation in speech perception resulted.

The adaptive algorithm implemented was based on the Griffiths-Jim ABF (Fig 1). Signals from two microphones are combined to form sum and difference signals. If we assume target speech directly in front of the patient and noise off to one side, then the speech signal is the same at both microphones, but the noise is different. The sum signal therefore contains signal plus noise components, but the difference signal only contains noise components (since the signal components from each microphone cancel). The difference signal is then filtered by an adaptive finite impulse response filter and subtracted from the sum signal. The coefficients of the filter are adjusted over time so as to minimize the total output power using a least mean squares criterion. If the noise is uncorrelated with the signal, only the noise component can be eliminated from the sum signal, thus leaving only the noise-free speech signal.

Note that if the speech signal is slightly off center, the difference signal also contains some speech so that cancellation of the speech signal can occur. To prevent this, we have modified the Griffiths-Jim algorithm by only updating the coefficients when we expect there is no speech signal present. This is done by comparing the time-averaged values of sum and difference signals; only when the difference signal is larger than half the sum do we update the coefficients. The rate of adaptation of the filter is dependent on filter length and signal power. Our tests were, however, more concerned with steady state performance. Informal measurements showed adaptation rates around 1 second. The length of the filter was set to 10 milliseconds. This relatively short filter length is a
compromise between power consumption and noise attenuation ability. It should be noted, however, that longer filter lengths may also cause adverse effects under more reverberant conditions.2

The algorithm was implemented on a modified Digital Sound Processor (DSP), which was originally developed for binaural cochlear implant work,6 but has since been adopted as our general research processor.7 The processor is based on the Motorola DSP56000 chip, and also includes two software-configurable radio frequency implant interfaces and an 8-channel A/D converter. To implement the ABF, sampling of left and right microphones was at an overall rate of 25 kHz. The total power consumption of the ABF processor was about 700 mW. When evaluating the ABF strategy, it was compared with a fixed broadside two-microphone strategy in which the signals from the two microphones were simply summed.

**MANIKIN TESTS**

In the physical evaluation, a Kemar manikin was placed 1.5 m from a loudspeaker in both a furnished soundproof booth and in a highly reverberant concrete stairwell. The direct-to-reverberant power ratio (D/R) was measured to be approximately 13 dB at 1.5 m in the soundproof booth and about 1.1 dB in the concrete stairwell. The reverberation times (T60) were measured by means of octave-spaced band-pass filters. The soundproof booth had a reverberation time near half a second, which is not atypical of living room conditions.8 For this reason we have called the test condition at 1.5 m from the speaker in the booth the "living room condition." The concrete stairwell had a reverberation time of about 2 seconds. A near-anechoic condition was furthermore tested by placing the manikin only 30 cm from the loudspeaker in the soundproof booth. In this condition, the D/R ratio was not measurable, since the contribution to the overall power by the reverberation in that case was very small.

Multitalker babble (Auditec, St Louis, Mo, catalog No. C146) was presented from the loudspeaker at 70 dB sound pressure level measured at the manikin's head. The manikin was then rotated while the noise attenuation was measured as a function of the angle of incidence of the noise. Under living room conditions, the attenuation of the noise when the manikin was turned more than 15° to 20° either side of the loudspeaker was approximately 10 dB. For comparison, the attenuation offered by the comparison scheme (in which the two microphones were added without the ABF) was about 3 dB at 90°. The effects of reverberation were clearly displayed by the results in the other acoustic environments: for the near-anechoic case, where the D/R is very high, as much as 20 dB attenuation of the noise was observed with the ABF. On the other hand, when tested in the concrete stairwell, where the reverberant field was almost as intense as the direct signal, only a 3-dB noise reduction was observed. This was as expected, since under highly reverberant conditions, we expected the coefficients of the adaptive filter to go to zero, so that the ABF algorithm reduces to the simple additive scheme without beamforming.

**PATIENT TESTS**

In order to test four cochlear implant patients, the ABF strategy was again compared with the simple two-microphone summation scheme. Both strategies were used as pre-processors for a Spectral Maxima Sound Processor (SMSp)9 electrical stimulation strategy. All four patients were regular users of an SMSp strategy. The patients were not allowed any take-home experience with the processor. Two test sessions were used to set up the processor so that maps were appropriate for each patient. None of the patients noted any significant qualitative differences in quiet between their own SMSP processor and the ABF processor. Open-set Australian SIT (Speech Intelligibility Test) sentences, described by Whitford et al.,10 were presented at 70 dB SPL to the patients from a speaker placed directly in front of them. Tests in noise additionally presented multitalker babble at 90° to the left and also at a level of 70 dB.

Results for each patient, as well as the average across the four patients, are shown in Fig 2 on the right and left sides, respectively. One-way analysis of variance comparing the two processing strategies, blocked by patients, was performed. Results showed significant improvements for speech perception in noise (p < .001) with the ABF processor as compared with the summation processor. No significant dif-

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**Table 1:**

<table>
<thead>
<tr>
<th>Signal-to-Noise Condition</th>
<th>ABF</th>
<th>SUM</th>
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<tr>
<td>Quiet 0 dB S/N</td>
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**Fig 1.** Schematic diagram of Griffiths-Jim adaptive beamformer. AFIR = adaptive finite response filter.

**Fig 2.** A: Average results for four cochlear implant patients with ABF and simple summation strategies, tested in living room conditions. A) Average results. B) Individual results. ABF = adaptive beamforming. SUM = summation.
ferences were shown for the tests in quiet, indicating that if any signal distortion resulted from the ABF, the effects were negligible. This is a good indication that our decision criterion for when to update filter coefficients was effective. These are encouraging results and also agree with the patients, who without exception commented that the speech perception task was much easier with the ABF processor than with the simple summation processor.

CONCLUSION

In summary, we have seen that a portable ABF can be of practical benefit to cochlear implant patients in modestly reverberant conditions, such as the home living room. Physical evaluation further showed that although large amounts of reverberation will reduce the ABF advantage, the ABF noise performance was never worse than the simple summation of two microphone signals. As newer DSP devices with lower power consumption become available, more microphones may also be added in portable ABF applications to allow simultaneous attenuation of multiple, spatially distinct interfering noise sources, and additional computation may be afforded, allowing some of the effects of reverberation to be reduced.

REFERENCES


BACK-TELEMETRY AND THE CLARION COCHLEAR PROSTHESIS

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From Advanced Bionics, Sarl, Habsheim, France (Zilberman), and Advanced Bionics Corporation, Sylmar, California (Santogrossi).

The Clarion cochlear prosthesis is an implanted electronic device that is designed to restore auditory sensation to deaf individuals. It incorporates a function known as back-telemetry. Telemetry, as it applies to cochlear implants, is the transmission of data from a speech processor to a receiver-stimulator (forward telemetry). The term back-telemetry refers to the radio frequency communication from the cochlear prosthesis to an external device that receives and interprets the information. Although communication in this direction is not strictly needed in the implementation of a cochlear implant system, it was decided, in the initial design of the Clarion cochlear prosthesis, to apply back-telemetry.

The fundamental reason behind this decision was the desire to develop a system that could emulate, as faithfully as possible, a percutaneous device. In 1985, it was evident that researchers using the Ineraid implant were taking advantage of their ability to measure certain properties of electrodes in the cochlea. The designers of the Clarion decided not to sacrifice this capability in their newly developed transcutaneous device. In addition to this principal objective, there were some concrete ideas about the added utility of this capability; these were primarily in three areas: 1) device diagnostics, 2) power consumption control mechanisms, and 3) adaptable coding strategies.

The design of the back-telemetry capability affected several components of the system. On the architectural level the instruction set of the communication link had to be expanded to allow an external device to control properties measured in the prosthesis that would be sent back to the external device. It also required establishing a second channel of communication that would coexist with the primary channel that would be responsible for sending information to the prosthesis. In order to comply with the architectural requirements, back-telemetry was implemented by means of a 10.7-MHz carrier, modulating digital information using frequency modulation. In the prosthesis itself, the back-telemetry capability required a multiplexer to switch to designated locations on the chip, analog-to-digital converters to encode the information, and a transmission subsystem that included the transmitter itself and an antenna. In addition, several switches were required to control which parameter was to be monitored at any given time. These switches were designed to be controlled from the outside in real time. The headpiece and the connecting cable were designed for bidirectional communication. Every external device that is part of the Clarion system was designed to make use of the back-telemetry by commanding it, receiving the information, and acting on it appropriately.

To date, back-telemetry is applied primarily to contribute to the safety and reliability of the device. This decision follows the company's concept of "safety and reliability first." Some applications of the back-telemetry include 1) a thorough prosthetic diagnostics as part of the device initialization, 2) an ongoing verification that the communication between the prosthesis and the external device is not interrupted, and
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