Conclusion

A totally implantable middle-ear system was designed using a microprocessor. The microprocessor can control the system using IR control signals and process the sound signal using FIR filter. The external control and recharge system was designed and power consumption of the system was reduced using on/off control by the microprocessor. A prototype system was developed according to the design specifications.

Acknowledgement

This study was supported by a grant of the Korea Health 21 R&D Project, Ministry of Health & Welfare, Republic of Korea. (02-PJ3-PG6-EV10-0001).

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Utilizing advanced hearing aid technologies as pre-processors to enhance cochlear implant performance

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Introduction

Although the common goals of both amplification devices are to enhance their users' speech understanding and listening comfort, especially in noise, and to improve the convenience of device use, the research and development of hearing aids and cochlear implants share little in common. Technology advances for hearing aids include:

1 directional microphones to reduce noise interference and improve speech understanding in noise (Cord MT etal., 2002; Hawkins and Yacullo, 1984; Killion et al., 1998)

2 in-situ microphone matching algorithms to optimize and maintain directional performance
3 adaptive directional microphones to automatically detect and reduce noise from different directions (Ricketts and Henry, 2002)
4 second-order directional microphones to further reduce noise interference
5 noise reduction algorithms to enhance listening comfort and speech understanding (Alcantara et al. 2003; Johns et al., 2001; Schum and Pogash, 2002)
6 feedback suppression algorithms to increase headroom and reduce feedback and occlusion
7 active and programmable telecoils to reduce noise interferences and to accommodate individual listening needs
8 automatic switches to switch between telecoil and microphone modes, between directional and omni-directional modes and among listening programs
9 laser shell making technologies.

Technology advances for cochlear implants, on the other hand, have greatly focused on the miniaturization of the speech processor, electrode array mechanics and speech-coding strategies. Most of the advanced features available in hearing aids are not widely available in cochlear implants. If they are offered, they are often in a less sophisticated form. For example, among the three cochlear implant manufacturers, only one (Cochlear Corporation) offers first-order directional microphones. However, these cochlear implants cannot be switched to omni-directional mode, which allows better detection of warning signals from behind and is less noisy in quiet or windy environments (Thompson, 1999; Ricketts et al., 2003). Further, the same manufacturer is the only one to offer telecoils yet the telecoils are not switchless, active or programmable.

Objectives

Before hearing-aid signal processing technologies can be widely implemented to enhance cochlear implants, it is necessary to explore if the utilization of hearing-aid technologies that were proven to enhance speech understanding and listening comfort for hearing aid users can be used as pre-processors to speech processors to enhance the speech understanding and listening comfort for cochlear implant users. Directional microphones have been reported to enhance speech understanding for hearing aid and cochlear implant users (Cord MT etal., 2002; Hawkins and Yacullo, 1984; Killion et al., 1998; Wouters J & Vanden Berghe, 2001), and noise reduction algorithms to improve listening comfort for hearing aid users (Alcantara et al. 2003; Johns et al., 2001; Schum and Pogash, 2002). The purpose of this study was to determine the feasibility of using hearing aid directional microphones and noise-reduction technologies as front-end processors to improve speech understanding and ease of listening for cochlear implant users.

Materials and methods

The CID recording of the NU6 monosyllabic words in speech spectrum noise were presented to a Knowles Electronic Manikin for Acoustic Research (KEMAR) wearing a pair of in-the-ear digital hearing aids. The speech was presented at 0(azimuth and uncorrelated noises was presented at 0°, +/–67.5°, +/–112.5° and +/–157.5° azimuths. The signal-to-noise ratios were set to 0 dB for subjects with normal hearing and +3 dB for subjects with cochlear implants and hearing aids. The hearing aids were programmed to:

1 omni-directional microphone
2 directional microphone
3 directional microphone plus noise reduction.

Other advanced features of the hearing aids were disabled. Under each condition, the hearing aids were programmed to be linear to avoid 'double compression' from both hearing aids and cochlear implants. The frequency response of the Om condition was programmed to be relatively flat when the hearing aids were worn in the KEMAR's ears.

The hearing aid-processed speech was recorded in Zwislocki couplers and then presented to subjects with cochlear implants, hearing aids and normal hearing. Subjects listened to the testing materials at their comfortable listening levels. Objective speech recognition tests and subjective rankings were conducted. The presentation order of the experimental conditions and the word lists were randomized.

Results

Repeated measure ANOVA indicated significant signal processor effect ($p < 0.001$) for all listeners with significantly higher speech recognition scores using Dm or DN than Om ($p < 0.0167$, post Hoc Tukey Kramer test) but no significant difference between Dm and DN ($p > 0.05$).The averaged improvement with the directional microphone was 11.7% points for cochlear implant, 21.5% for hearing aid and 23.7% for normal-hearing listeners.

All subjects ranked Om as the most difficult and DN as the easiest with the exception of two normal-hearing subjects who commented that Om was the most difficult, and Dm and DN were similar in ease of listening. The ranking differences among the signal processors were significant using the Friedman two-way ANOVA for all subject groups ($p < 0.05$).

Conclusions

The results of this exploratory study were encouraging. Further studies on different signal processors and different technologies are needed to increase generalizability of the application. The long-term goal of this project is to utilize hearing-aid signal processors as pre-processors to speech processors to enhance cochlear implants, so that cochlear implant manufacturers do not need to reinvent the technologies that are already available in hearing aids, and cochlear implant users can take advantage of the advanced features as soon as these features are available in hearing aids. (Chung et al., submitted).

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Sound separation in noise and competing voice with normal-hearing subjects

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Objective

'Auditory stream' was a concept first introduced by Bregman and Campbell (1971) to describe auditory perception of a series of sounds from background noise. The auditory perceptual process has been identified to be governed by both the primitive cues (spectral, temporal and spatial cues) and the schema-based processes (central auditory perception). In this study, two experiments were conducted to investigate the use of primitive cues by people with normal hearing with two masking signals, broadband noise and competing voice.

Three hypotheses were examined:

1 Two simultaneous sounds presented diotically (two concurrent sounds presented into two ears together) will be more difficult to separate than the same two sounds presented dichotically (two concurrent sounds presented into two ears separately).
2 Two simultaneous sounds with generally similar spectral and temporal characteristics will be more difficult to separate than grossly different sounds.
3 The dichotic presentation of speech to one ear and speech-shaped noise in the opposite ear has no effect on thresholds for the speech signal.

Materials and methods

The experimental conditions and testing procedures were modified from the two experiments designed by Blamey et al. (2001). The subjects in this research were eight normally hearing adults.

Experiment 1: Separation of a voice and a noise

The speech stimuli used in experiment 1 were closed-set spondees recorded by a female Australian English speaker. The starting level was –30 dB (re. 65 dB HL audiometer