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Utilizing hearing aid directional microphones and noise reduction algorithms to improve speech understanding and listening preferences for cochlear implant users

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Abstract. The purpose of this study was to investigate if applying hearing aid directional microphones and noise reduction algorithms as preprocessors to cochlear implant speech processors can enhance speech understanding and listening preference of cochlear implant users. Hearing aid preprocessed speech materials were recorded when KEMAR was wearing a pair of nine-channel digital hearing aids and sitting at the center of an eight-speaker array. The hearing aids were programmed to omnidirectional microphone (Om), directional microphone (Dm), and directional microphone plus noise reduction algorithms (Dm+NR). Cochlear implant users listened to the hearing aid preprocessed speech materials in sound field to simulate the use of hearing aids as preprocessors to cochlear implants. They repeated the words and ranked their listening preferences of the three experimental conditions. Results indicated that the Dm+NR condition yielded better speech recognition and higher listening preferences than the Dm condition which, in turn, was better than the Om condition ($p<0.01$) for cochlear implant users. These results indicate that hearing aid directional microphones and noise reduction algorithms can be used to enhance speech understanding and listening preferences of cochlear implant users. © 2004 Elsevier B.V. All rights reserved.

Keywords: Hearing aid; Cochlear implant; Speech recognition; Directional microphone; Noise reduction

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1. Introduction

In the last several years, we have observed the greatest leap in the advancement of hearing aid signal processing technologies. Many of these technologies can potentially be applied to cochlear implants to increase speech understanding, sound quality and convenience of cochlear implant use. These technologies include directional microphones, microphone matching algorithms, adaptive directionality algorithms, second-order directional microphones, noise reduction algorithms, active and programmable telecoils, and automatic switches for different microphone modes, different input modes or listening programs. The purpose of this study was to investigate if hearing aid directional microphones and noise reduction algorithms can be used as front-end processors to improve speech understanding and listening preferences for cochlear implant users. If the results are positive, it is possible that other hearing aid signal processing technologies can also be used to enhance cochlear implants.

2. Methods

Hearing aid preprocessed speech in noise materials were presented to subjects with cochlear implants to simulate the use of hearing aid directional microphone and noise reduction algorithms as front-end processors to cochlear implants. Subjects with normal hearing were included as controls.

2.1. Hearing aid settings

A pair of nine-channel digital hearing aids was made for Knowles Electronic Manikin for Acoustic Research (KEMAR) and the hearing aid output was recorded in the Zwislocki coupler by a pair of ER-11 1/2″ microphones. The hearing aids were programmed to three experimental conditions:

(1) omnidirectional microphone (Om),
(2) directional microphone (Dm), and
(3) directional microphone plus noise reduction (Dm+NR).

Under each condition, the hearing aids were programmed to be linear with flat frequency responses (i.e., ±3 dB between 300 and 6000 Hz). The directional microphones had fixed directionality and the noise reduction algorithms utilized strategies to take advantage of the spectral separation between speech and noise in multiple frequency channels. The reverberation time of the recording room was approximately 500 ms.

2.2. Recording of speech testing materials

W-22 monosyllabic word lists were presented at 0° azimuth to KEMAR wearing the pair of digital hearing aids and sitting at the center of an eight-speaker array. Equal amplitude and uncorrelated speech spectrum noise was presented at 0°, ±67.5°, ±112.5° and ±157.5° azimuths. The speech in noise materials was recorded at signal-to-noise ratios of +3 and 0 dB for subjects with cochlear implants and normal hearing, respectively. The speech envelope of the phrase “Say the word stove” recorded at signal-to-noise ratio of +3 dB is shown in Fig. 1.

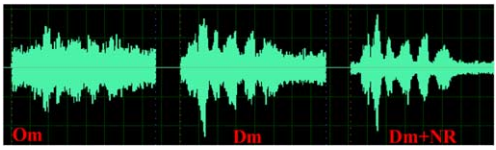

Fig. 1. The phrase "Say the word stove" processed by the hearing aids under the three experimental conditions.

2.3. Subjects

Sixteen subjects with cochlear implants (n=8) and normal hearing (n=8) participated in this research study. Among the subjects with cochlear implants, two wore Clarion CII, two wore Nucleus 24 SPrint SP and four wore Nucleus 24 Esprit 3G speech processors. The hearing sensitivity of the subjects with normal hearing was within 20 dB HL from 250 to 8000 Hz in octave intervals.

2.4. Testing procedures

Subjects listened to the recorded speech materials in sound field at their comfortable listening levels. They repeated the words for the speech recognition test. Then they ranked their listening preferences. Subjects with cochlear implants used their preferred listening program during tests.

3. Results

The average speech recognition scores for each experimental condition using phonemic scoring are shown in Fig. 2. Repeated measure ANOVA indicated significant subject group and signal processor effects (p<0.01). Post hoc tests indicated that the Dm+NR condition yielded significantly better speech recognition scores than the Om and Dm conditions (p<0.01) and Dm was better than Om for subjects with cochlear implants (p<0.01). For subjects with normal hearing, the Dm and Dm+NR conditions yielded significantly better scores than the Om condition (p<0.05).

The average rankings of listening preferences are shown in Fig. 3. Repeated measure ANOVA indicated significant signal processor effect (p<0.01). Post hoc tests indicated

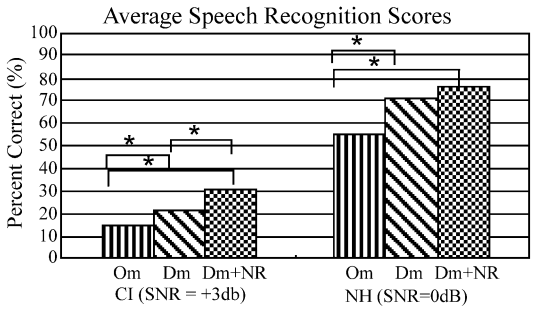

Fig. 2. Average speech recognition scores of subjects with cochlear implants (CI) and normal hearing (NH).

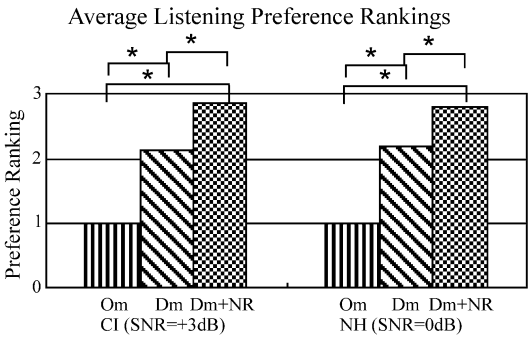

Fig. 3. Average rankings of listening preferences (3=most preferable, 2=middle, and 1=least preferable).

that the Dm+NR conditions were more preferable than the Dm conditions ($p<0.01$) which, in turn, were more preferable than the Om conditions ($p<0.01$) for all subject groups.

4. Discussions

The results of this study indicated that hearing aid directional microphones and noise reduction algorithms, can be used as preprocessors to help cochlear implant users understand speech in background noise. Subjectively, cochlear implant users preferred listening to speech processed by the hearing aid preprocessor.

One of the cautions in applying hearing aids as preprocessors to cochlear implants is that the effectiveness of algorithms may differ among different hearing aid manufacturers. Chung et al. [1] explored the effectiveness of directional microphones and noise reduction algorithms of a six-channel digital hearing aid and reported no significant differences between the directional microphone condition and the directional microphone plus noise reduction condition in speech understanding (difference=1%). In this study, a pair of nine-channel digital hearing aids was used and the speech recognition score for Dm+NR condition was significantly higher (9%) than the Dm condition for cochlear implant users.

Another caution is that the lack of improvement in speech recognition for subjects with normal hearing does not always imply insignificant improvement for cochlear implant users. In this study, the speech recognition scores for subjects with normal hearing were not significantly different for the Dm and Dm+NR conditions, yet the difference was significant for cochlear implant users. Hochberg et al. [2] also reported similar findings using another noise reduction algorithm. The disparity may attribute to the inherent differences in how the signals are encoded in the normal hearing ears and in the cochlear implant speech processors. It is possible that the noise reduction algorithms enhanced the modulation of speech in background noise (Fig. 1) and enabled the speech processor to pick the speech peaks with higher accuracy and thus enhanced speech understanding for the cochlear implant users.

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