

# Using hearing aid directional microphones and noise reduction algorithms to enhance cochlear implant performance

**King Chung**

*Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN  
kingchung@purdue.edu*

**Fan-Gang Zeng**

*Department of Otolaryngology-Head and Neck Surgery, University of California, Irvine, CA  
fzeng@uci.edu*

**Susan Waltzman**

*Department of Otolaryngology, New York University School of Medicine, New York, NY  
susan.waltzman@med.nyu.edu*

**Abstract:** Hearing aids and cochlear implants are two major hearing enhancement technologies but yet share little in research and development. The purpose of this study was to determine whether hearing aid directional microphones and noise reduction technologies could enhance cochlear implant users' speech understanding and ease of listening. Digital hearing aids serving as preprocessors were programmed to omni-directional microphone, directional microphone, and directional microphone plus noise reduction conditions. Three groups of subjects were tested with the hearing aid processed speech stimuli. Results indicated that hearing aids with directional microphones and noise reduction algorithms significantly enhanced speech understanding and listening comfort.

© 2004 Acoustical Society of America

**PACS numbers:** 43.66.Ts, 43.64.Me, 43.72.-p, 43.71.Ky [BLM]

**Date Received:** October 22, 2003      **Date Accepted:** January 16, 2004

## 1. Introduction

The common goals of both hearing aids and cochlear implants are to enhance users' speech understanding and listening comfort, especially in noise, and to improve the convenience of device use. To achieve these goals, hearing aid and cochlear implant manufacturers have gone through two markedly different paths. In the past several years, technology advances for hearing aids include but are not limited to (1) directional microphones to improve speech understanding in noise and reduce noise interference;<sup>1-5</sup> (2) *in situ* microphone matching algorithms to optimize and maintain directional performance;<sup>6</sup> (3) adaptive directionality algorithms to automatically detect and reduce noise from different directions;<sup>6,7</sup> (4) second-order directional microphones to further reduce noise interference;<sup>8,9</sup> (5) noise reduction algorithms to enhance listening comfort and speech understanding, especially in noise with narrow bandwidths;<sup>4,10-12</sup> (6) active and programmable telecoils to reduce interference and to accommodate individual needs;<sup>13</sup> and (7) automatic switches to switch between telecoil and microphone modes, between directional and omni-directional modes and among listening programs.

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.<sup>14-16</sup> It is interesting to note that most of the advanced features available in hearing aids are not widely available in cochlear implants and, if offered, are often offered in a less sophisticated form. For example, among all cochlear implant manufacturers, only one (Cochlear Cooperation) offers first-order directional microphones. Yet the cochlear implants cannot be switched to omni-directional microphones, which allow better detection of warning signals

from behind and are less noisy in quiet or windy environments.<sup>17</sup> In addition, the same manufacturer is the only one to offer telecoils but the telecoils are neither active nor programmable.

Directional microphones have been reported to enhance speech understanding for hearing aid and cochlear implant users<sup>1-5,18</sup> and noise reduction algorithms have been reported to improve listening comfort for hearing aid users.<sup>4,10,12</sup> The purpose of this study was to determine the feasibility of using hearing aid directional microphones and noise reduction technologies as preprocessors to improve speech understanding and ease of listening for cochlear implant users.

## 2. Methods

Speech testing materials were presented to a Knowles Electronic Manikin for Acoustic Research (KEMAR) wearing a pair of in-the-ear digital hearing aids. The processed speech was recorded in Zwislocki couplers and then presented to three groups of subjects.

### 2.1 Hearing aid settings

The digital hearing aids had six signal processing channels, first-order directional microphones, *in situ* microphone matching algorithms, and noise reduction algorithms. The noise reduction algorithms constantly monitored the amount of modulation in each signal processing channel. As the envelope of speech is normally modulated but the envelope of noise is not, the noise reduction algorithms reduced the gain of the channels at which unmodulated signals dominated and let signals with modulation pass.<sup>19,20</sup> Other advanced features, e.g., adaptive directionality, voice-activated noise reduction algorithms, adaptive feedback suppression, and automatic switches, were not tested and thus deactivated.

The hearing aids were programmed to three experimental conditions:

- (1) omni-directional microphone (Om),
- (2) directional microphone (Dm), and
- (3) directional microphone plus noise reduction (DN).

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 (i.e.,  $\pm 3$  dB between 400 and 5000 Hz) when the hearing aids were worn in the KEMAR’s ears. The gain of the hearing aids was set to 10 dB for the Om condition. The differences in articulation index weighted directivity index (AIDI) between the Om and Dm conditions estimated by the Directional Hearing Aid Analyzer at 32 locations<sup>21,22</sup> were 2.6 dB for the left and 2.7 dB for the right hearing aids.

### 2.2 Recording of speech testing materials

The Central Institute for the Deaf (CID) recordings of the NU6 monosyllable word lists were recorded when KEMAR was wearing the hearing aids and sitting at the center of an eight-speaker array. The reverberation time of the recording room was approximately 500 ms. Loudspeakers 1 to 7 were located at  $0^\circ$ ,  $\pm 67.5^\circ$ ,  $\pm 112.5^\circ$ , and  $\pm 157.5^\circ$  azimuths. They generated an even and uncorrelated noise field at an overall level of 80 dB SPL. Loudspeaker 8 was designated for speech presentation at  $0^\circ$  azimuth. It was calibrated to produce levels of 80 or 83 dB SPL for recordings of speech testing materials at signal-to-noise ratios (SNRs) of 0 or +3 dB, respectively. A calibration noise was recorded for each experimental condition when all eight speakers were active simultaneously.

To record speech at 0-dB SNR, the recording level of the computer sound card was fixed in order to capture the differences in overall levels among the experimental conditions. The relative levels of the phrase “Say the word limb” processed under Om, Dm, and DN conditions were 0, -8.1, and -14.7 dB, respectively, for the left channel and 0, -9.1, and -16.5 dB, respectively, for the right channel. To record speech at +3-dB SNR, the recording level of the sound card was adjusted so that the calibration noise for each experimental condition peaked at the same recording level. This arrangement was made to reduce the need for subjects with

Table 1. The cochlear implant information of subjects with cochlear implants.

Subject	Ear	CI brand	CI model	Strategy and rate	Years of CI use
S1	L	Nucleus 24	ESPrIt 3G	ACE, 900 Hz	1.5
S2	R	Nucleus 24	SPrInt SP	ACE, 720 Hz	6
S3	R	Clarion	CII	CIS, 812.5 Hz	1
S4	L	Nucleus 24	SprInt SP	ACE, 720 Hz	3.5

cochlear implants and hearing aids to adjust their volume controls. The speech testing materials were recorded at 44.1-kHz sampling rate and with 16-bit resolution.

### 2.3 Subjects

Table 1 summarizes the cochlear implant information of subjects with cochlear implants ( $N=4$ ). The post-fitting sound field hearing thresholds for the subject who used the Clarion cochlear implant were between 25 and 30 dB HL from 250 to 4000 Hz in octave intervals and between 40 and 50 dB HL for the three subjects who used the Nucleus 24 cochlear implants. Subjects with normal hearing ( $N=4$ ) and hearing aids ( $N=4$ ) were included as controls for audibility and signal processing effects. The hearing thresholds of subjects with normal hearing were less than or equal to 20 dB HL from 250 to 8000 Hz in octave intervals. Subjects with hearing aids had sensorineural hearing loss and their unaided thresholds were between 20 and 65 dB HL from 500 to 4000 Hz.

### 2.4 Testing procedures

All testing was carried out either in the recording room or in a sound proof chamber (ICA 101146). Subjects with normal hearing listened to all experimental conditions monaurally at a SNR of 0 dB via the earphones of a discman (Panasonic SL-S150). The discman was used so that subjects could easily adjust the volume to their comfortable listening levels. If subjects prefer listening with the left ear, the left channel of the recording was played and vice versa.

Subjects with hearing aids listened to speech testing materials recorded at a SNR of +3 dB via sound field when they were wearing their own digital hearing aids. Their hearing aids did not have directional microphones, noise reduction algorithms, or volume controls and were set to the NAL-NL1 recommended prescription based on subjects' individual hearing sensitivity. The presentation level was set at either 70 or 75 dB SPL depending on the subject's preference.

Subjects with cochlear implants listened to hearing aid processed speech at a SNR of +3 dB. The Om condition was presented via a loudspeaker in sound field in order to simulate cochlear implants with omni-directional microphones being used in real life. The Dm and DN conditions were presented via direct audio input (DAI) in order to simulate hearing aids being used as preprocessors to cochlear implant speech processors and the electric outputs of the hearing aids being fed into the speech processors via DAI. The speech recognition scores generated from these conditions can be compared directly because the subjects sat within the critical distance during sound field testing (i.e., no reverberation effect) and Zeng and Gavin<sup>23</sup> showed little difference in speech intelligibility between loudspeaker and DAI conditions.

Subjects used the same listening program throughout the experiment. They adjusted the volume control of their cochlear implants, if needed, at the beginning of each experimental condition so that speech was presented at comfortable listening levels.

For *speech recognition tests*, subjects listened to the hearing aid processed word lists with the carrier phrase "Say the word \_\_\_". They were asked to repeat the last word of the phrase and they were encouraged to guess. The presentation order of the experimental conditions and the word lists were randomized.

For *ease of listening ranking*, subjects were asked to rank the three experimental conditions (1=easiest, 2=middle, 3=most difficult) at the end of the speech recognition test as if

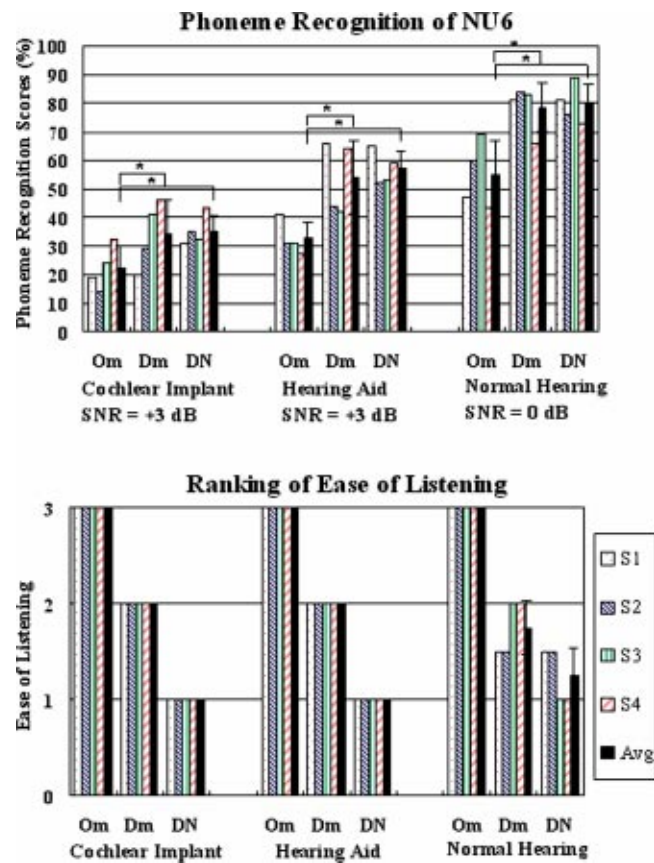


Fig. 1. (a) Phoneme recognition scores and (b) ease of listening rankings (1=easiest, 2=middle, 3=hardest) of speech processed by the digital hearing aids.

they were three different hearing aids or cochlear implant programs that they were going to wear in real life.

**3. Results**

Figure 1a shows speech recognition scores and 1b shows ease of listening rankings for each subject. Repeated measure ANOVA indicated significant signal processor effect ( $p < 0.001$ ). All subject groups showed 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%, 21.5%, and 23.7% for subjects with cochlear implants, hearing aids, and normal hearing, respectively.

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 by ranks for all subject groups ( $p < 0.05$ ).

**4. Discussions**

The results of the present study showed that directional microphones enhanced speech understanding, and that the addition of noise reduction algorithm significantly improved listening comfort of cochlear implant users. These results are generally consistent with that obtained in hearing aid users.<sup>1-6,10,12</sup> The reason for the lower speech recognition scores compared to the scores obtained in some recent studies using sentence materials in quiet<sup>15,24-27</sup>

may be because NU6 is a set of monosyllabic word lists which are especially difficult for cochlear implant users when presented at a low SNR and in a uniform speech spectrum noise field.

The positive results of this exploratory study generated interests in new applications of hearing aid technologies, yet many questions remain unanswered. Specifically, hearing aid manufacturers often employ different signal processing technologies to compensate for hearing loss with cochlear origin and modern advanced digital hearing aids are implemented with very different parameters, e.g., number of channels, filter band slopes, compression, adaptive dynamic range optimization algorithm,<sup>25</sup> to name a few. It is unsure if there were interactions between hearing aid signal processors and cochlear implant speech processors due to the differences in signal processing technologies and coding strategies. In addition, the proprietary algorithms offered by different manufacturers may be implemented using different techniques even if they meant to have similar functions. It is unknown if they would be equally effective when combined with cochlear implants with different coding strategies.

Should further research show the use of hearing aid signal processing technologies as preprocessors for cochlear implants desirable, the most efficient way to take advantage of advanced hearing aid technologies would be to incorporate these technologies/algorithms into the cochlear implant speech processor chips. If hearing aid manufacturers cannot disclose their proprietary algorithms, it is possible that an in-the-ear earpiece can be built to house the hearing aid signal processors, then the hearing aid processed signal can be fed into cochlear implant speech processors via direct connection or wireless transmission. A third way would be to integrate hearing aid signal processor chips into cochlear implant speech processors.

Regardless of the technical implementation methods, the advantages of marriage between hearing aid and cochlear implant technologies are at least twofold. Clinically, it ensures timely delivery of any current or future advanced signal processing technologies to cochlear implant users to enhance their speech understanding and listening comfort as well as convenience of cochlear implant use. Scientifically, sharing technologies will save valuable research and development costs so that hearing aid and cochlear implant manufacturers can develop future generation of technologies to benefit both hearing aid and cochlear implant users.

### Acknowledgments

We would like to thank Deirdre Knight for data collection and Dr. Arlene Neuman for providing equipment and facilities for making recordings and data collection, and for her valuable inputs to the draft of this manuscript.

### References and links

- <sup>1</sup>D. B. Hawkins and W. S. Yacullo, "Signal-to-noise ratio advantage of binaural hearing aids and directional microphones under different levels of reverberation," *J. Speech Hear Disord.* **49**, 278–286 (1984).
- <sup>2</sup>M. C. Killion, R. Schullien, L. Christensen, D. Fabry, L. Revit, P. Niquette, and K. Chung, "Real world performance of an ITE directional microphone," *Hear. J.* **51**, 24–38 (1998).
- <sup>3</sup>T. Ricketts, G. Lindley, and P. Henry, "Impact of compression and hearing aid style on directional hearing aid benefit and performance," *Ear Hear.* **22**(4), 348–361 (2001).
- <sup>4</sup>B. E. Walden, R. K. Surr, M. T. Cord, B. Edwards, and L. Olson, "Comparison of benefits provided by different hearing aid technologies," *J. Am. Acad. Audiol.* **11**(10), 540–560 (2000).
- <sup>5</sup>J. Wouters, J. V. Berghe, and J. B. Maj, "Adaptive noise suppression for a dual-microphone hearing aid," *Int. J. Audiol.* **41**(7), 401–407 (2002).
- <sup>6</sup>G. Matsui and T. Lemons, "A special report on new digital hearing instrument technology," *Hear. Rev.* **8**(4), 7–31 (2001).
- <sup>7</sup>T. A. Ricketts and P. Henry, "Evaluation of an adaptive directional-microphone hearing aid," *Int. J. Audiol.* **41**, 100–112 (2002).
- <sup>8</sup>T. A. Powers and V. Hamacher, "Three-microphone instrument is designed to extend benefits of directionality," *Hear. J.* **55**(10), 38–45 (2002).
- <sup>9</sup>W. Soede, F. A. Bilsen, A. J. Berkhout, and J. Verschuure, "Directional hearing aid based on array technology," *Scand. Audiol. Suppl.* **38**, 20–27 (1993).
- <sup>10</sup>J. L. Alcantara, B. C. Moore, V. Kuhnel, and S. Launer, "Evaluation of the noise reduction system in a commercial digital hearing aid," *Int. J. Audiol.* **42**(1), 34–42 (2003).
- <sup>11</sup>M. Johns, V. Bray, and M. Nilsson, "Effective noise reduction," *Audiology Online* (2001), <http://audiologyonline.com>.

- <sup>12</sup>D. Schum, "Noise-reduction circuitry in hearing aids: Goals and current strategies," *Hear. J.* **56**(6), 32–40 (2003).
- <sup>13</sup>B. Marshall, "Advances in technology offer promise of an expanding role for telecoils," *Hear. J.* **55**(9), 40–41 (2002).
- <sup>14</sup>J. M. Ostroff, E. A. David, D. B. Shipp, J. M. Chen, and J. M. Nedzelski, "Evaluation of the high-resolution speech coding strategy for the Clarion CII cochlear implant system," *J. Otolaryngol.* **32**(2), 81–86 (2003).
- <sup>15</sup>M. W. Skinner, L. K. Holden, L. A. Whitford, K. L. Plant, C. Psarros, and T. A. Holden, "Speech recognition with the nucleus 24 SPEAK, ACE, and CIS speech coding strategies in newly implanted adults," *Ear Hear.* **23**(3), 207–223 (2002).
- <sup>16</sup>B. S. Wilson, C. C. Finley, D. T. Lawson, R. D. Wolford, D. K. Eddington, and W. M. Rabinowitz, "Better speech recognition with cochlear implants," *Nature (London)* **18**, 352(6332), 236–238 (1991).
- <sup>17</sup>S. C. Thompson, "Dual microphones or directional-plus-omni: Which is best?" *High Perform. Hear. Solut.* **3**, 31–35 (1999).
- <sup>18</sup>J. Wouters and J. Van den Berghe, "Speech recognition in noise for cochlear implantees with a two-microphone monaural adaptive noise reduction system," *Ear Hear.* **22**(5), 420–430 (2001).
- <sup>19</sup>B. W. Edwards, C. J. Struck, P. Dharan, and Z. Hou, "New digital processor for hearing loss compensation based on the auditory system," *Hear. J.* **51**(9), 44–52 (1998).
- <sup>20</sup>M. Latzel, J. Kiessling, and S. Margolf-Hackl, "Optimizing noise suppression and comfort in hearing instruments," *Hear. Rev.* **10**(3), 76–82 (2003).
- <sup>21</sup>A. Neuman, K. Chung, M. Steele, and M. Bakke, "A newly developed in-situ measurement system: The Directional Hearing Aid Analyzer," *Int'l Hear Aid Conference, Lake Tahoe, CA* (2002).
- <sup>22</sup>K. Chung and A. C. Neuman, "Measurements of In-Situ Hearing Aid Directionality in A Real Room," presentation at Am. Aud. Soc. Meeting, Scottsdale, AZ (2003).
- <sup>23</sup>F.-G. Zeng and J. J. Galvin, "Amplitude compression and phoneme recognition in cochlear implant listeners," *Ear Hear.* **20**, 60–73 (1999).
- <sup>24</sup>B. L. Fetterman and E. H. Domico, "Speech recognition in background noise of cochlear implant patients," *Otolaryngol.-Head Neck Surg.* **126**(3), 257–263 (2002).
- <sup>25</sup>C. J. James, M. W. Skinner, L. F. Martin, L. K. Holden, K. L. Galvin, T. A. Holden, and L. Whitford, "An investigation of input level range for the nucleus 24 cochlear implant system: speech perception performance, program preference, and loudness comfort ratings," *Ear Hear.* **24**(2), 157–174 (2003).
- <sup>26</sup>T. T. Valimaa, M. J. Sorri, and H. J. Lopponen, "The effect of a multichannel cochlear implant on phoneme perception," *Scand. Audiol. Suppl.* **52**, 51–53 (2001).
- <sup>27</sup>P. J. Blamey, G. J. Dooley, C. J. James, and E. S. Parisi, "Monaural and binaural loudness measures in cochlear implant users with contralateral residual hearing," *Ear Hear.* **21**, 6–17 (2000).