# Effects of directional microphone and adaptive multichannel noise reduction algorithm on cochlear implant performance

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Although cochlear implant (CI) users have enjoyed good speech recognition in quiet, they still have difficulties understanding speech in noise. We conducted three experiments to determine whether a directional microphone and an adaptive multichannel noise reduction algorithm could enhance CI performance in noise and whether Speech Transmission Index (STI) can be used to predict CI performance in various acoustic and signal processing conditions. In Experiment I, CI users listened to speech in noise processed by 4 hearing aid settings: omni-directional microphone, omni-directional microphone plus noise reduction, directional microphone, and directional microphone plus noise reduction. The directional microphone significantly improved speech recognition in noise. Both directional microphone and noise reduction algorithm improved overall preference. In Experiment II, normal hearing individuals listened to the recorded speech produced by 4- or 8-channel CI simulations. The 8-channel simulation yielded similar speech recognition results as in Experiment I, whereas the 4-channel simulation produced no significant difference among the 4 settings. In Experiment III, we examined the relationship between STIs and speech recognition. The results suggested that STI could predict actual and simulated CI speech intelligibility with acoustic degradation and the directional microphone, but not the noise reduction algorithm. Implications for intelligibility enhancement are discussed. © *2006 Acoustical Society of America.* [DOI: 10.1121/1.2258500]

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## I. INTRODUCTION

Although the performance of cochlear implant (CI) users has been increasing constantly with recent improvements in CIs (Loizou, 1998; Zeng, 2004), understanding speech in noise still remains a great challenge. Multiple attempts have been made to increase the signal-to-noise ratio (SNR) of sounds before the signal reaches the CI preprocessor but with limited success. For example, the Audallion BEAMformer that summed the inputs from two directional microphones worn in the implanted ear and in the contralateral ear was reported to have limited benefit on localizing the source of sounds (Figueiredo *et al.*, 2001), limited directional effects (Goldsworthy, 2005), and limited acceptability among CI users. In addition, several groups of researchers have reported positive findings using spectral subtraction noise reduction algorithms to enhance the speech recognition of CI users in background noise (Hochberg *et al.*, 1992; Weiss, 1993; Goldsworthy, 2005). The computational demand, however, prevents such algorithms from being implemented in wearable CIs. As technologies advance, new generations of front-end processors (e.g., directional microphones and adaptive directional microphones) are implemented in some recent CI models to combat noise but few studies have reported the effectiveness of these strategies. In this study, we will explore the effects of two noise reduction strategies commonly used in hearing aids, directional microphones and adaptive multichannel (AMC) noise reduction algorithms, on cochlear implant performance.

Directional microphones have higher sensitivity to sounds coming from the front than from the sides or the back. They are applied to hearing devices to take advantage of spatial separation between speech and noise. Most directional microphones implemented in high performance hearing devices utilize two omni-directional microphones, which are equally sensitive to sounds from all directions. When the directional microphones are activated, the electrical signal generated by the back microphone is subtracted from that of the front microphone. Depending on the ratio of the distance between the two omni-directional microphones and the electronic delay added to the back microphone, the polar pattern can vary from bipolar, to hypercardioid, supercardioid or cardioid. This means that the least sensitive location(s) of the microphone (i.e., the null) can change from 90° and 270° in bipolar patterns to 180° in cardioid pattern. The overall improvement provided by directional microphones is roughly 3–5 dB in real-world environments with low reverberation compared to omni-directional microphones for listeners with

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acoustic hearing (Amlani, 2001; Chung, 2004; Ricketts, 2001; Valente *et al.*, 2000; Wouters *et al.*, 1999; Bentler, 2005).

Speech is a highly modulated signal (Plomp, 1983; Rosen, 1992). Speech sounds are characterized by temporal modulations between 2 and 50 Hz with a peak modulation rate of 3–8 Hz depending on speaking rate (Powers *et al.*, 1999). Noise, however, is often displayed as having a steadier temporal envelope or having a modulation rate outside the modulation range of speech. When speech and noise coexist in a signal, the modulation depth of speech is reduced because noise fills the silence between phonemes or sentences and masks the low-level components of speech.

Most noise reduction algorithms implemented in digital hearing devices utilize an adaptive gain reduction mechanism to take advantage of spectral separation between speech and noise across multiple frequency channels (i.e., AMC noise reduction algorithms). They use a modulation detector to monitor the modulation characteristics of the signal within each frequency channel and apply gain reduction to reduce noise interference. If a high modulation depth with center modulation rates similar to those of speech is detected in a channel, the noise reduction algorithm infers that speech exists in the channel and the SNR is high so the gain of that frequency channel is not reduced. Otherwise, the algorithms infer that the SNR is low or noise dominates the frequency channel and the gain of the channel is reduced. Usually the amount of gain reduction increases as the SNR decreases. The exact amount of gain reduction depends on the decision rules of the particular noise reduction algorithm which, in turn, is determined by the modulation depth, the estimated SNR in the frequency channel, overall level of the signal, the level detected in the channel, the frequency-importance of the channel for speech recognition, and so on (Bentler, 2005; Chung, 2004; Ricketts and Hornsby, 2005). Although few noise reduction algorithms are reported to be effective in modulated background noise (e.g., single-talker babble), many are reported to reduce noise interference and increase sound quality, listening comfort, or overall preference in noise with limited temporal variations (e.g., speech spectrum noise) (Edwards *et al.*, 1988; Walden *et al.*, 2000; Johns *et al.*, 2003; Kuk *et al.*, 2002; Mueller *et al.*, 2003; Powers and Hamacher, 2002; Ricketts and Hornsby, 2005).

The speech transmission index (STI) was first proposed to predict speech intelligibility in rooms with different acoustic properties (Houtgast and Steeneken, 1973). It is calculated by comparing the changes in modulation depths of the different frequency regions between a probe (a modulated noise) and a transmitted signal after the probe is transmitted across an acoustic medium. A number of researchers have also proposed several speech-based transmission index calculation methods to utilize speech as the probe signal, instead of the modulated noise (Payton *et al.*, 2002; Drullman *et al.*, 1994; Ludvigsen *et al.*, 1990; Koch, 1992; Holube and Kollmeier, 1996). The advantage of speech-based calculation methods is that they allow the estimation of speech intelligibility in different acoustic environments and under different speech signal processing algorithms. In general, the higher the modulation depth in the transmitted or processed signal, the higher the speech transmission index and the higher the predicted speech intelligibility. The disadvantage is that STI has been reported to fail to predict the speech intelligibility of nonlinearly processed speech for people using acoustic hearing (e.g., for spectral subtraction noise reduction algorithm, see Ludvigsen *et al.*, 1993, Goldsworthy and Greenberg, 2004; for envelope clipping, see Drullman, 1995; for compression, see Hohmann and Kollmeier, 1995; for envelope thresholding, see Goldsworthy, 2005). It appeared that factors other than the modulation depth affected the performance of people using acoustic hearing.

Nevertheless, there are fundamental differences between how speech is encoded in acoustic and electric hearing. Previous studies have shown that listeners with normal hearing and hearing aid users can use both spectral and temporal information for speech understanding (Van Tasell *et al.*, 1987, 1992). Yet, CI users are forced to rely on the temporal envelope cues to understand speech because the spectral fine structure information is only coarsely presented in 6 to 22 channels (Loizou, 1998). Our previous recordings showed that both directional microphones and AMC noise reduction algorithms increase temporal modulation depths of speech in background noise (Chung *et al.*, 2004b). It is possible that both of these strategies make speech envelopes more salient and can help the CI speech processor determine which speech peaks to present to the electrodes and thus improve speech recognition for CI users. Additionally, as the STI calculations are also based on temporal modulations in nonoverlapping filter bands, similar to CIs, it is possible that STI could be used to predict speech intelligibility of both noise reduction strategies for CI users.

In a previous study, Chung *et al.* (2004a, b) conducted a series of preliminary studies to investigate whether directional microphones and noise reduction algorithms could enhance CI performance. They recorded speech in noise testing materials processed by a 9-channel and a 6-channel digital hearing aid when the hearing aids were set to omnidirectional microphone, directional microphone, and directional microphone plus AMC noise reduction. The testing materials were then presented to CI users via direct audio input. The results showed that the two conditions with the directional microphone yielded significantly better speech recognition and higher sound quality rankings than the omnidirectional microphone condition for both hearing aids (Chung, 2004a, b). Significantly better speech recognition scores were also observed for the directional microphone with noise reduction condition compared with the directional microphone alone condition for the 9-channel digital hearing aid. These studies, however, were conducted with relatively small sample sizes (i.e., CI users $N=4$ and 8). It is unknown if these positive results can be generalized to a larger CI population. Additionally, the effect of noise reduction algorithms alone was not investigated and the CI users showed near floor performance in some conditions because the speech testing materials were recorded at a low SNR (i.e., +3 dB).

The purposes of this series of experiments were to investigate whether (1) directional microphones and AMC noise reduction algorithms used as preprocessors could enhance CI performance in noisy environments; and (2) a speech-based STI could be applied to predict the CI performance using these two noise reduction strategies. These research questions were addressed in 3 experiments: in Experiment I, the speech recognition ability of CI users was tested when listening to speech in noise testing materials processed by a digital hearing aid with directional microphone and noise reduction algorithm. The CI users also rated their overall preferences of the test conditions at three SNRs in a paired-comparison paradigm. In Experiment II, the speech recognition ability of two groups of listeners with normal hearing was tested when they listened to the same testing materials with 4- or 8-channel CI simulations. In Experiment III, the relationship between the speech recognition scores of CI users and normal hearing individuals and STIs calculated using the speech-based STI method proposed by Payton *et al.* (2002) was explored. In all the experiments involving research participants, repeated measure designs were used to reduce variability due to subject variability and all participants were blinded to the testing conditions to eliminate systematic errors caused by subject bias.

## II. EXPERIMENT I

A custom in-the-ear 9-channel digital hearing aid was made for Knowles Electronic Manikin for Acoustic Research (KEMAR)'s right ear. Speech testing materials were recorded when the hearing aid was programmed to different settings then presented to CI users via direct audio input. This procedure was used to simulate the condition in which a hearing aid signal processor preceded a CI speech processor.

As CI users have a wide range of speech understanding ability, we recorded the speech testing materials at five SNRs to avoid floor and ceiling effects, and to bracket the SNR for 50% correct speech recognition (SNR_50). All the recordings were made in an anechoic chamber to minimize the effects of reverberation on STI calculations in Experiment III. As one of the goals of amplification devices is to improve perceived sound quality, CI users also rated their overall preference of the processed speech recorded at three SNRs. The following are detailed descriptions of the experimental procedures used in this study:

### A. Methods

#### 1. Subjects

Seven male and 13 female CI users (mean age=58.2 years old) were recruited for this experiment. Their demographic information and information on their CIs are summarized in Table I. All listeners participated in the speech recognition in noise test and 13 listeners rated overall preferences of the experimental conditions. The tests were conducted at Purdue University in West Lafayette, IN and at University of California in Irvine, CA.

#### 2. Characteristics of the digital hearing aid

The digital hearing aid used in this study had a first-order directional microphone with a fixed hypercardioid pattern. Only one hearing aid was used because we wanted to control the amount of directional effects in the directional condition across subjects. The test hearing aid had a −3 dB/octave low-frequency roll-off in the directional setting compared to the omni-directional setting in order to compensate for half of the low-frequency roll-off of the first-order directional microphone.

TABLE I. The demographic and cochlear implant information of listeners.

| Subject | Gender | Age | Test ear | Number of years of CI use | Speech processor | Coding strategy |
|---|---|---|---|---|---|---|
| 1 | M | 79 | L | 2;3 | ESPrit 3G | CIS |
| 2 | M | 47 | R | 11;10 | Spectra 22 | SPEAK |
| 3 | M | 73 | R | 1;11 | Auria | HiRes |
| 4 | F | 41 | L | 4;9 | Combi40+ | CIS |
| 5 | F | 60 | R | 6;7 | Clarion | CIS |
| 6 | F | 72 | R | 4;10 | ESPrit 3G | ACE |
| 7 | F | 43 | R | 19;6 | Ineraid | CIS |
| 8 | F | 62 | R | 1;0 | ESPrit 3G | ACE |
| 9 | M | 78 | R | 1;10 | ESPrit 3G | ACE |
| 10 | M | 63 | L | 14;10 | Spectra 22 | SPEAK |
| 11 | F | 67 | R | 4;3 | Clarion | MPS |
| 12 | F | 72 | L | 11;0 | S-Series | CIS |
| 13 | M | 27 | L | 0;6 | ESPrit 3G | ACE |
| 14 | F | 44 | L | 5;11 | ESPrit 3G | SPEAK |
| 15 | F | 51 | L | 6;7 | S-Series | SAS |
| 16 | M | 57 | R | 3;7 | ESPrit 3G | ACE |
| 17 | F | 57 | R | 6;0 | ESPrit 3G | SPEAK |
| 18 | F | 45 | L | 1;9 | Combi40+ | CIS |
| 19 | F | 70 | R | 7;7 | ESPrit 3G | ACE |
| 20 | F | 55 | L | 3;0 | ESPrit 3G | ACE |

The noise reduction algorithm implemented in this hearing aid was an AMC noise reduction algorithm with nine signal processing channels. The amount of gain reduction in each channel was inversely proportional to the estimated SNR. No gain reduction was executed if the SNR was estimated to be at or higher than 24 dB. A maximum of 12 dB gain reduction was exercised if the SNR was estimated to be at 0 dB at a frequency channel. The noise reduction algorithm reduced the noise to within 3 dB of the steady noise level in 8 s and to within 1 dB in 14 s.

#### 3. Preparation of speech recognition testing materials

Prior to making the recordings of the speech testing materials, the hearing aid was programmed to have linear signal processing and flat frequency response when it was worn in KEMAR's ear (i.e., flat *in situ* frequency response). The expansion algorithm at very low input level was turned off in all testing conditions. Thus the omni-directional microphone setting of the hearing aid provided little frequency or amplitude alterations to the incoming sounds. It acted as a reference condition for other hearing aid processed conditions as if no hearing aid preprocessor were added to the CI speech processor.

The speech recognition testing materials were recorded using the equipment setup in Fig. 1. In the calibration process, Computer 1 (2.39 GHz Pentium 4 with 1 Gbyte of RAM) presented the speech spectrum calibration noise from the Hearing In Noise Test (HINT, Nilsson *et al.*, 1994) to Speaker 1, which was a Mackie HR824 powered amplifier

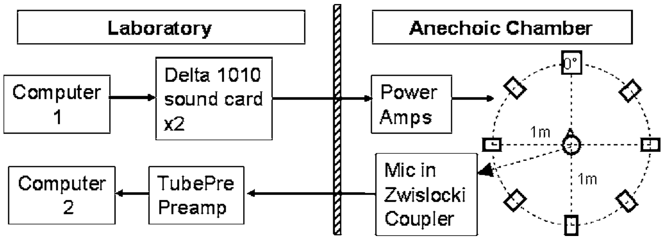


FIG. 1. The equipment setup for the recording of hearing aid processed testing materials.

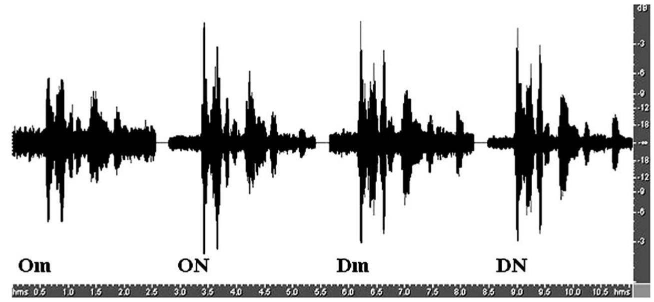


FIG. 2. The temporal envelopes of the sentence "The house had nine bedrooms" processed under four hearing aid settings: omni-directional microphone (Om), omni-directional microphone plus noise reduction algorithms (ON), directional microphones (Dm), and directional microphones plus noise reduction algorithms (DN) at SNR of +8 dB.

with a ±1.5 dB frequency response from 39 to 20 kHz. The level of the calibration noise was adjusted to be 68, 70.5, 73, 75.5, and 78 dB SPL measured by a sound level meter placed in the center of the speaker array in the absence of KEMAR.

An uncorrelated uniform noise field was generated by a total of 8 speakers at 0°, 45°, 90°, 135°, 180°, 225°, 270°, and 315° (Speakers 2–8 were Hafler M5 speakers). The level of the speech spectrum calibration noise was adjusted to be 56 dB SPL from individual speakers which resulted in a uniform noise field with an overall level of 65 dB SPL. An uncorrelated noise field was used to avoid the comodulation masking release effect which is a result of correlated noise field and could yield an increase in speech understanding scores compared to an uncorrelated noise field (Cox and Bisset, 1984; Grose and Hall, 1992; Kwon, 2002; Moore, 1990). A continuous noise field, instead of the gated noise used in the original HINT test, was utilized to ensure that the noise reduction algorithm was always engaged.

In the recording process, KEMAR was placed in the center of the speaker array. The output of the hearing aid was recorded by an ER11 $\frac{1}{2}$ in. microphone (Etymotic Research) placed in the medial opening of a Zwislocki coupler attached to KEMAR's ear canal, and then fed to Computer 2 (1.8 GHz Intel Pentium M processor with 512 Mbytes RAM).

Four lists of HINT sentences were recorded at each SNR (i.e., +3, +5.5, +8, +10.5, and +13 dB) when the hearing aid was set to omni-directional microphone (Om), omni-directional microphone with noise reduction (ON), directional microphone (Dm), and directional microphone with noise reduction (DN). The sentences were presented approximately 10 s after the presentation of noise to ensure that they were recorded after the actions of the noise reduction algorithm had stabilized. Each sentence was separated by approximately 5–6 s of noise, as in the original HINT test.

After the testing materials were recorded, the rms levels of speech were equalized across experimental conditions in order to minimize the possibility that CI speech processors with narrow input-dynamic range peak-clipping signals with higher levels. The temporal envelope plots of the sentence "The house has nine bedrooms" processed by the four hearing aid settings is shown in Fig. 2.

Three sentences were arbitrarily chosen at each of the +3, +8, +13 dB SNR to create paired-comparison tokens for overall preference ratings. Each sentence formed 12 tokens (i.e., 6 combinations: Om-ON, Om-Dm, Om-DN, ON-Dm, ON-DN, Dm-DN; and 6 reversals: ON-Om, Dm-Om, DN-Om, Dm-ON, DN-ON, DN-Dm). A 500 ms silence was inserted between the sentences. All wave-editing tasks were carried out using Adobe AUDITION 1.0.

### 4. Speech recognition in noise test

Prior to the administration of speech recognition tests, the key words in the HINT sentence lists were analyzed and the comfortable listening levels were determined for individual listeners. Any auxiliary words for continuous tense were not counted as key words, but the same words used as verb were counted. For example, the word "is" was not counted as a key word in "(A/The) *car* (*is/was*) *going too fast*," but it was counted as a key word in "(A/The) *fire* (*is/was*) *very hot*." Articles were not counted as key words. Any variations allowed in the original HINT test were also allowed in this study. For example, in the sentence "(A/The) *fire* (*is/was*) *very hot*," both "is" and "was" were counted as correct. Using this scoring method, the number of key words for the sentence lists ranged from 38 to 45 words. In addition, each listener's comfortable listening level for the recorded speech via direct audio input was determined using the loudness scale and procedures developed by the Independent Hearing Aid Fitting Forum (IHAFF, 1994, i.e., "1" – very soft, "2" – soft, "3" – comfortable but slightly soft, "4" – comfortable, "5" – comfortable but slightly loud; "6" – loud but ok, and "7" – too loud).

During speech recognition tests, the sentences recorded at +8 dB SNR were administered to the listeners first. If a listener obtained a speech recognition score close to 0% or 100% at SNR of +8 dB for most hearing aid settings, HINT lists recorded at +3 and +13 dB were presented to see if the listener reached the floor or ceiling of performance. If so, no further tests were administered and the listener was discharged. Otherwise, HINT lists from higher and lower SNRs were presented to bracket the SNR_50 for the individual CI user. These procedures were adopted because a three-parameter sigmoidal function was used to fit the data points and SNR_50s were estimated using the parameters generated from the sigmoidal function. Thus, if a listener was tested at their floor or ceiling performance levels, the sigmoidal function would generate erroneous results. The speech testing materials were presented to listeners at their comfortable listening levels (i.e., "4" in the IHAFF scale). Listener 13 lis-

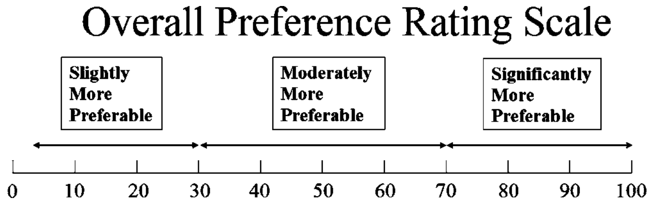


FIG. 3. The overall preference rating scale used in the paired comparison categorical rating paradigm.

tened at a "comfortable but slightly soft" level (i.e., "3") because she reported distortions of speech at the "4" level during a routine check.

The percentage correct speech recognition scores were calculated by dividing the number of key words the listener repeated correctly by the total number of key words in the particular sentence list. The percent correct scores for each hearing aid signal processing setting were then converted to SNR_50s using a three parameter sigmoidal function:

$$\text{Percent Correct} = A/(1 + \exp(-(\text{SNR} - X0)/B), \tag{1}$$

$$\text{SNR\_50} = X0 - B \ln(A/0.5 - 1), \tag{2}$$

where $A$ is the asymptotic performance, $X0$ is the SNR at which the percent correct performance is 50% of $A$, and $B$ is a parameter related to the slope.

#### *5. Overall preference ratings*

Thirteen CI users rated their overall preferences of sentences in a combined paired comparison and categorical rating paradigm. After listening to each paired-comparison token, they reported their preference of Condition 1 or Condition 2 to the examiner and then rated the magnitude of their preferences using the scale shown in Fig. 3. A total of 18 tokens (i.e., 6 combinations/reversals $\times 3$ sentences) were administered at each SNR to each listener.

In the scoring process, the 6 combinations and their reversals were grouped into 6 pairs (e.g., Om-ON tokens were grouped with ON-Om tokens). If a listener indicated that they preferred Condition 1 (say ON) over Condition 2 (say Om) by 30 points, a score of 30 was entered for ON and a score of 0 was entered for Om. The average of Om or ON equaled the sum of scores in the Om-ON combination divided by the total number of trials.

### B. Results

#### *1. Speech recognition in noise*

The raw speech recognition scores of all subjects are depicted in Fig. 4. The SNR_50 for each hearing aid setting was calculated for all the listeners except listeners 1, 6, 7, 15, and 18 whose SNR_50s could not be calculated in at least one hearing aid setting because of poor performance. Subsequently, the SNR_50 of 16 listeners was analyzed using repeated measure ANOVA on hearing aid settings to determine whether the directional microphone and noise reduction algorithm could improve listeners' speech recognition in background noise. The results showed significant main effects of hearing aid settings ($F[15,3]=17.9, p<0.0001$). The average SNR_50 for the four hearing aid conditions were 6.0, 4.4, 2.5, and 1.9 dB for the Om, ON, Dm, and DN conditions.

Scheffe pairwise comparisons were carried out to determine hearing aid settings that yielded significant differences. The difference was significant between Om and Dm, Om and DN, ON and Dm, and ON and DN ($p<0.0083$, adjusted to account for multiple test conditions). The critical difference for significance was 1.9 dB. The absence of significant difference between Om and ON or between Dm and DN indicated that the directional microphone improved CI users' speech understanding in noise but the noise reduction algorithm did not.

The CI users exhibited a wide range of speech coding and electrical stimulation strategies. While it was not the intention of this study to test the applicability of the directional microphone and the noise reduction algorithm to a particular group of CI users, the results of the largest group of participants with the same speech processor (i.e., the seven EsPrit 3G listeners) were analyzed as a group to investigate whether the speech processor played a role in the process. The repeated measure ANOVA indicated a significant hearing aid setting main effect ($F[6,3]=6.13, p<0.01$). Scheffe pairwise comparison tests showed significant difference between Om and Dm, and between Om and DN. The critical difference for significance was 3.1 dB. The general results were similar to those of the whole group, that directional microphone enhanced speech understanding while the noise reduction algorithm did not.

#### *2. Overall preference ratings*

Six paired t-tests were performed to determine significant differences among the comparison pairs at each SNR. The $p$ level for 0.05 significance level was adjusted to be 0.003 to account for multiple t-tests [i.e., $0.05/(3\text{ SNR} \times 6\text{ tests})$]. The average overall preference ratings of the comparison pairs are summarized in Fig. 5 and the significantly different pairs are indicated by asterisks ($^*$). Significant results were obtained between Om-ON, Om-Dm, Om-DN, ON-DN, and Dm-DN at SNRs of +3 and +8 dB ($p<0.0025$). The magnitude of preferences ranged from 23% (slightly more preferable) to 57% (moderately preferable). Overall, DN was ranked the most preferable and Om the least preferable. No significant differences were reported between ON and Dm at these SNRs. These results indicate that, at low SNRs, CI users preferred the conditions with the noise reduction algorithm and/or the directional microphone over Om, and their preferences were similar for the conditions with noise reduction alone and with directional microphone alone.

At a SNR of +13 dB, significant differences were obtained between Om-DN and between ON-DN only. The magnitudes of the preferences were 45% (moderately preferable) for both comparison pairs. The differences between all other pairs did not reach statistical significance. These results suggest CI users only preferred a combination of directional microphone and noise reduction algorithm over the unprocessed or noise reduction conditions at a high SNR. This may be because speech was already clear to them and enhance-

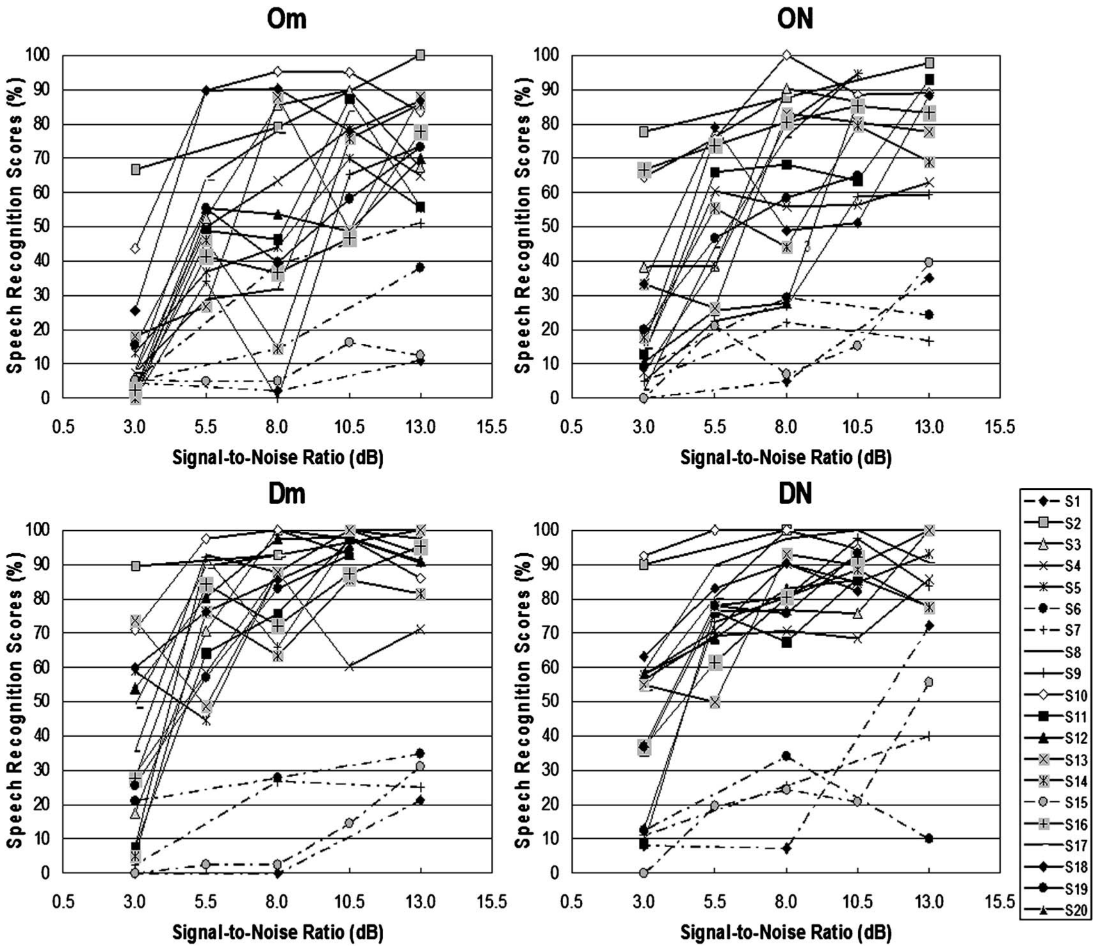


FIG. 4. The raw scores obtained by all the cochlear implant listeners when they listened to speech in noise materials processed by a digital hearing aid set to omni-directional microphone (Om), omni-directional microphone plus noise reduction algorithms (ON), directional microphones (Dm), and directional microphones plus noise reduction algorithms (DN) at 5 SNRs. Om represents a condition in which no hearing aid signal processing was added to the cochlear implant speech processor.

ments by directional microphone or noise reduction algorithm alone did not make a significant impact on their overall preferences.

### C. Discussion

CI users obtained significantly better speech recognition scores when they listened to speech processed by directional microphones (i.e., Dm and DN) compared to the conditions without directional microphones (i.e., Om and ON, respectively). The directional microphone provided an average of 3.5 and 3.6 dB improvement in SNR (Om vs Dm) for all the CI users and the EsPrit 3G listeners, respectively. The amount of improvement is consistent with that reported in studies with simulated real-world environments (Amlani,

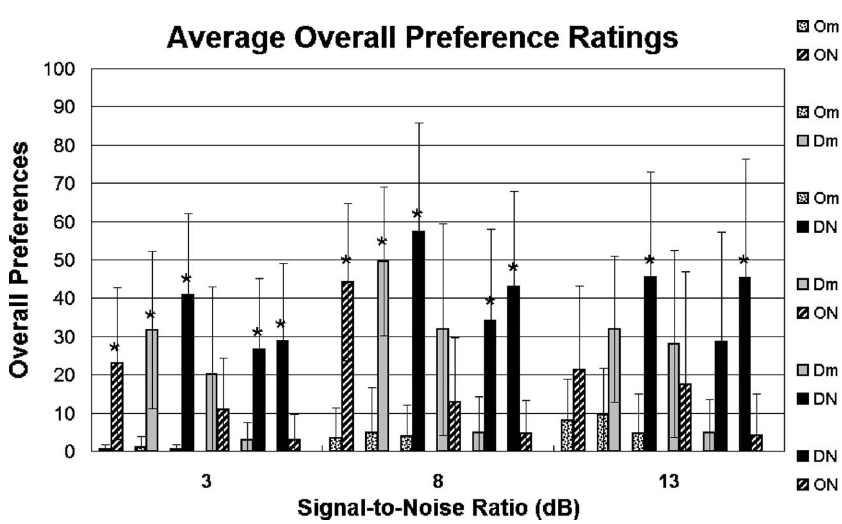


FIG. 5. The average overall preference ratings of cochlear implant listeners when they listened to speech processed at omni-directional microphone (Om), omni-directional microphone plus noise reduction algorithms (ON), directional microphones (Dm), and directional microphones plus noise reduction algorithms (DN) at 3 SNRs.

2001; Chung, 2004; Ricketts, 2001; Valente *et al.*, 2000; Wouters *et al.*, 1999; Bentler, 2005). Note that noise was not only presented from the sides and the back but also from 0° azimuth in the present study. This is rarely the case in other studies. The improvement from the directional microphone could be higher if noise was only presented from the sides and the back.

On the other hand, the SNR_50 obtained in conditions with AMC noise reduction algorithm (i.e., ON or DN) did not reach statistical significance when compared to that in the conditions without the noise reduction algorithm (i.e., Om and Dm, respectively). The difference between Om and ON for all CI users and the EsPrit listeners was 1.6 and 2.4 dB, respectively. It seems that the noise reduction algorithm used in this study somewhat increased the speech recognition of CI users. Future investigations are needed to explore if there are interaction effects between the noise reduction algorithm and speech processors, and if a different amount of noise reduction or noise reduction algorithms from other digital hearing aids would have a better noise reduction effect in helping CI users understand speech in background noise.

In Chung *et al.* (2004b), the improvement provided by the DN condition reached statistical significance when the CI users were tested at a SNR of +3 dB. In this study, the DN condition did not provide any significant improvement compared to Dm for either the CI or the EsPrit 3G group. This suggested that improvement provided by DN was minimal when the noise reduction algorithm was applied across a wider range of SNRs.

Subjectively, CI users preferred conditions with directional microphone and/or noise reduction algorithm. This result is consistent with other reports in hearing aid literature on the same hearing aid for listeners with acoustic hearing (Bray and Nilsson, 2001; Johns *et al.*, 2003). The amount of preference ranged from slightly more preferable to moderately preferable. According to Gabrielsson and Sjogren (1979), sound quality is a multidimensional phenomenon, namely, clarity, sharpness, brightness, fullness, spaciousness, nearness, noisiness, and loudness. The weightings of these dimensions are different for individual listeners and different tasks. Although it is unclear which dimension(s) determined subjective preferences in this study, it is possible that the reduced overall noise contributed to the higher overall preferences of the conditions with noise reduction algorithms (see Fig. 2). The implications are that AMC noise reduction algorithm can be applied in CIs to enhance perceived sound quality and to reduce the overall background noise.

Another interesting finding is that the average preference ratings of this group of CI users suggest that the preference ratings obtained using categorical rating paradigm were not transitive. In other words, if the listeners rated condition $A$ $x$ amount more preferable than condition B, and condition $B$ $y$ amount more preferable than condition $C$, the amount of preference for condition C compared to condition A did not equal to $x+y$. For example, at SNR of +3 dB, the difference between Om and DN rated by the listeners was 40.3. If transitivity held, the sum of differences (in absolute values) for Om_ON and Om_Dm (i.e., improvement from noise reduction (Om_ON)+ improvement from directional microphone (Om_Dm)=sum of improvement (Om_DN)) should be 40.3. However, the sum of differences was 53.0. Similar lack of transitivity was also observed in SNR of +8 and +13 dB. The implication is that, if we desire to know the preference ratings between multiple experimental conditions using paired comparison categorical rating paradigm, the ratings should be performed but not inferred or calculated.

## III. EXPERIMENT II

Previous studies have shown that simulating listening to the CI for normal hearing individuals is a viable tool for providing insight into the effect of various signal processing strategies on CI users. At the same time, it eliminates sources of variability such as differences in electrical stimulation strategies, survival of cell bodies of first-order auditory neurons, location of electrodes relative to surviving neurons, etc. (Dorman and Loizou, 1998; Fu *et al.*, 1998, 2004; Stickney and Zeng, 2004; Nelson and Jin, 2004). In this study, normal hearing individuals listened to the speech materials recorded in Experiment I which were then processed to simulate CI processing with 4 or 8 channels of temporal envelope cues.

### A. Materials and Methods

#### *1. Subjects*

Two groups of listeners with normal hearing were recruited ($N=27$) to participate in the study. Their hearing thresholds from 250 to 8000 Hz were tested in octave intervals prior to admission in the study. The hearing sensitivity of the listeners was within 20 dB HL at all test frequencies and they had normal middle ear functions.

Group I [NH(Mod4)] consisted of 15 listeners who listened to 4-channel CI simulation. The data for 3 listeners were excluded in the final analysis because their scores in one or more hearing aid settings (mainly Om and ON) were so poor that SNR_50 could not be estimated. Subsequently, 12 listeners with normal hearing were recruited in Group II [NH(Mod8)] to listen to the 8-channel CI simulated speech. The mean ages for the final Group I and Group II listeners were 21.3 and 20.7 years old, respectively.

#### *2. Speech recognition in noise for CI simulations*

The speech testing materials recorded in Experiment I were processed by a MATLAB program based on the algorithms used in the experiments conducted by Shannon *et al.* (1995). This program extracted and preserved the temporal envelope cues of the speech sentences to 4 or 8 channels and, at the same time, eliminated spectral fine structures within the channels. Briefly, the MATLAB program divided the stimuli into four or eight spectral filter bands by using bandpass filters. The cut-off frequencies of these bandpass filters were calculated from the Greenwood map (1990), which was intended to divide the tonotopically arranged basilar membrane into equal distances and map the corresponding physical frequency range accordingly. The final wave form was derived from the sum of the temporal envelopes after each channel of filtered stimuli was full-wave rectified, low-pass

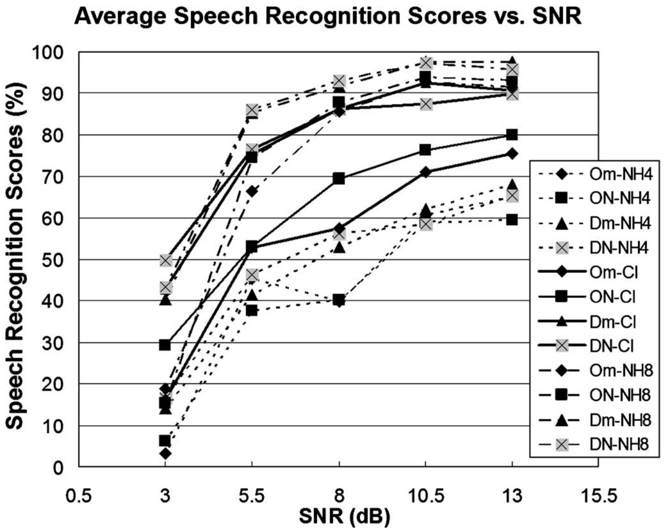


FIG. 6. The average speech recognition scores of cochlear implant users and normal hearing individuals listening to 4 and 8 channels of cochlear simulations in omni-directional microphone (Om), omni-directional microphone plus noise reduction algorithms (ON), directional microphones (Dm), and directional microphones plus noise reduction algorithms (DN) conditions at 5 SNRs.

filtered at 400 Hz, multiplied by a white noise, and then filtered by its corresponding filter again.

During testing, the CI simulated speech materials were presented from a computer to a GSI 61 Clinical Audiometer (Grason-Stadler, Inc). Listeners with normal hearing listened to the stimuli from a pair of ER-3A insert earphones at 65 dB HL. Ten CI simulated sentences in quiet were given for practice prior to testing. A verbal feedback of the correct response was given after they listened to the practice sentences but not during actual testing.

### B. Results

The average scores of the two groups of normal hearing individuals and the 16 CI users are plotted in Fig. 6 and the standard deviations are tabulated in Table II. The average scores of CI users fell between those of the NH(Mod4) and NH(Mod8) groups and the standard deviations of the CI users were relatively higher than those of the normal hearing users. It is also noteworthy that the speech recognition scores obtained in Dm and DN for CI users were equal to or exceeded the scores obtained in Om and ON for the NH(Mod8) group.

The percent correct scores of normal hearing individuals obtained at different SNRs were converted to SNR_50s. A repeated measure ANOVA was also performed on hearing aid settings for each listener group. No significant main effect was found for the NH(Mod4) listeners ($p > 0.05$). However, a significant main effect of hearing aid settings was found for the NH(Mod8) listeners ($F[15,3]=18.3, p < 0.0001$).

The average SNR_50s for the NH(Mod8) were 4.5, 4.4, 3.4, and 3.3 dB for Om, ON, Dm, and DN. Scheffe pairwise comparisons were carried out to examine significant differences among the hearing aid settings for the NH(Mod8) group. Significant differences were found between Om and Dm, Om and DN, ON and Dm, ON and DN ($p < 0.0083$).

TABLE II. The standard deviations of speech recognition scores of CI users (CI) and normal listeners listening to 8 (Mod8) and 4 (Mod4) channels of cochlear implant simulation.

| Group | SNR | | | | |
|---|---|---|---|---|---|
| | 13.0 | 10.5 | 8.0 | 5.5 | 3.0 |
| | | | Om | | |
| Mod(8) | 9.2% | 6.2% | 8.2% | 13.4% | 11.8% |
| CI | 20.1% | 21.6% | 30.8% | 22.2% | 14.3% |
| Mod(4) | 12.4% | 11.6% | 17.5% | 16.1% | 3.8% |
| | | | ON | | |
| Mod(8) | 5.7% | 3.9% | 7.2% | 13.2% | 11.3% |
| CI | 17.4% | 21.8% | 27.2% | 20.7% | 21.3% |
| Mod(4) | 12.1% | 9.2% | 8.3% | 18.0% | 4.2% |
| | | | Dm | | |
| Mod(8) | 2.6% | 4.1% | 6.9% | 11.3% | 9.9% |
| CI | 20.4% | 23.2% | 23.8% | 24.8% | 26.2% |
| Mod(4) | 14.2% | 19.2% | 10.9% | 10.9% | 6.8% |
| | | | DN | | |
| Mod(8) | 5.9% | 3.8% | 6.5% | 10.4% | 16.3% |
| CI | 13.8% | 19.7% | 18.2% | 18.6% | 25.1% |
| Mod(4) | 14.2% | 16.4% | 19.3% | 10.5% | 7.6% |

The critical difference was 0.8 dB. No significant difference was found between Om and ON or between Dm and DN.

### C. Discussion

The results of this experiment indicate that the directional microphone enhanced the speech recognition ability of normal hearing individuals in noise when they listened to 8-channel CI simulation, but the noise reduction algorithm did not. These results were consistent with those obtained for the CI users in Experiment I. The speech recognition scores of this study were also consistent with previous research studies that the performance of CI users fell roughly between the performances of normal hearing individuals listening to 4 and 8 channels of CI simulated speech in noise (Friesen *et al.*, 2001; Stickney and Zeng, 2004; Zeng *et al.*, 2005). One exciting finding is that when CI users listened to speech processed by Dm and DN, their speech recognition scores exceeded or equaled those of the NH(Mod8) listeners in the Om and ON conditions, especially at low SNRs.

The results of NH(Mod4) showed no significant difference between any hearing aid setting for normal hearing individuals when they listened to 4-channel CI simulation. Previous studies showed that the performance of normal hearing individuals listening to 4- and 6-channel CI simulated speech was similar to the performance of CI users with an equal number of speech processing channels (Fu *et al.*, 1998; Dorman *et al.*, 1997; Dorman and Loizou, 1998). This suggests that the extra spectral information provided by the 8-channel simulation made it a more sensitive tool for detecting changes in signal processing strategies for this mixed group of multichannel CI users whose speech processors have more than 4 signal processing channels.

## IV. EXPERIMENT III

In this experiment, the applicability of a speech-based STI program to predict the speech intelligibility of speech processed by a directional microphone and an AMC noise reduction algorithm was explored. The calculated STIs were also used as indications of the amount of improvement in temporal envelope modulations in the processed signal to shed light on the factors that determine speech intelligibility.

### A. Materials and Methods

#### *1. Speech-based STI calculation method*

There are at least four originally proposed speech-based STI programs, namely the normalized covariance method by Koch (1992) and Holube and Kollmeier (1996), envelope regression method by Ludvigsen *et al.* (1990), real cross-power spectrum method by Drullman *et al.* (1994), and magnitude cross-power spectrum method by Payton *et al.* (1994, 1999, 2002). These methods differ in how the transmission indexes, or the modulation depth of each frequency band, are estimated.

The speech-based STI calculation method proposed by Payton *et al.* (2002) was used in this study because Goldsworthy and Greenberg (2004) reported that this method produced STI values that are closest to those calculated by the original non-speech-based STI by Hougast and Steeneken (1985) if the transmitted signal was degraded in acoustic environments. The STI was calculated from the transmission indexes of speech at seven filter bands centered at 125, 250, 500, 1000, 2000, 4000, and 8000 Hz multiplied by the speech weighting of the corresponding frequency band. In this study, the probe was a sound file with 30 concatenated HINT sentences in quiet (i.e., the reference sound file) and the transmitted signals were 20 speech-in-noise sound files processed by Om, ON, Dm, and DN at SNR of +3, +5.5, +8, +10.5, and +13 dB (i.e., the processed sound files).

#### *2. Recording of processed sound files*

New recordings of the processed signals with concatenated sentences were made because the recordings from Experiment I had noise between sentences. Before the recording of the processed signals, a 300 ms 1000 Hz tone was placed 700 ms before and after the first and last sentences in the reference sound file. These tones served as markers to mark the beginning and the end of the concatenated sentence stream. The same calibration and recording procedures used in Experiment I were carried out to record the processed sound files. After the recordings were made, the marker tones and the silence were removed from the reference sound file. In the processed sound file, the noise before the first marker, the first marker tone, and the 700 ms noise were removed from the beginning of the sentence stream, and the 700 ms and the second marker tone were removed from the end of the sentence stream. These procedures generated a reference and processed signals with exact length. The STIs were then calculated by comparing the modulation depth in the reference and the processed files.

### B. Results

Three-parameter sigmoidal functions were used to fit the STIs and speech recognition scores obtained in the Om and Dm conditions for the three groups of listeners (Fig. 7). The

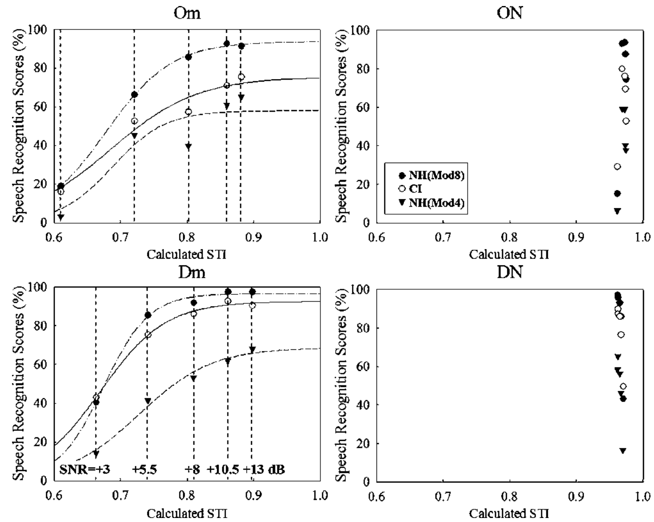


FIG. 7. The relationship between the average speech recognition scores and the calculated speech transmission indexes (STIs) at the four hearing aid settings for the three groups of listeners at 5 SNRs.

speech recognition scores increased monotonously with STIs, indicating that STI could be applied to predict the speech recognition scores of CI users and normal hearing individuals in these conditions.

No attempt was made to fit the scores obtained in the ON and DN conditions because the STI values for these conditions only differ by 0.0129. Fitting the data based on such a small range on one variable may bear little practical relevance and lead to erroneous conclusions.

### C. Discussion

The monotonous increase in STI from low to high SNR conditions in the Om and Dm condition indicated that the modulation depth of the temporal envelopes of the processed sentences increased as the SNR increased (see the dashed lines in Fig. 7). The rapid increase in STIs at low SNRs and slower increase at high SNRs indicates that the STI predicts a greater improvement in speech recognition scores for a reduction in noise at a low SNR than the same reduction in noise at a high SNR. In this study, the speech recognition scores were in general agreement with the STI prediction in the Om and Dm conditions. Therefore, we concluded that STI was a good predictor of acoustic degradations and directional microphones for CI users. These results were also consistent with findings of studies involving listeners with acoustic hearing (Steeneken and Houtgast, 1982; Payton *et al.*, 1994; Goldworthy, 2005; Ricketts and Hornsby, 2003).

In contrast, the ranges of calculated STIs were 0.0129 for ON and 0.0082 for DN across the SNRs while the ranges of the speech recognition scores varied between 40.1% and 77.8% for the three groups of listeners. These results in electrical hearing parallel the findings of previous studies in acoustic hearing that STI is a poor predictor of speech recognition scores for nonlinearly processed speech (Ludvigsen *et al.*, 1993; Drullman, 1995; Hohmann and Kollmeier, 1995; Goldsworthy, 2005). The extremely narrow range of STIs suggests that the modulation depth of the speech temporal envelope is almost identical across the SNRs. This finding is also consistent with the hearing aid manufacturer's descriptions of the noise reduction algorithm, which increased gain reduction as the level of noise increased. In other words, higher gain reductions were applied to the signal as the SNR decreased at frequency channels with noise dominance and thus the hearing aid output had similar temporal envelope modulations at all SNRs.

The relationships between STIs and speech recognition scores obtained in different acoustic and signal processing conditions suggests that one of the determining factors for speech understanding prediction is the within-channel SNR. Although both directional microphone and AMC noise reduction algorithm increased the modulation of temporal envelope of speech in background noise, only the directional microphone significantly enhanced the speech intelligibility of CI users and normal hearing individuals listening to CI simulated speech. As mentioned in Sec. I, directional microphones are more sensitive to sounds from the front than the sides or the back. When the testing materials were recorded with speech presented from the front and noise presented from all around, the SNR across frequency regions and within frequency channels was improved in the Dm condition compared to the Om condition.

The noise reduction algorithm, on the other hand, does not improve within-channel SNR because any gain reduction applied to the frequency channels would have affected both speech and noise. The fact that the speech recognition scores of ON and DN increased as SNR increased, instead of reaching a plateau as predicted by the calculated STIs, suggests that one of the determining factor for speech recognition is the within-channel SNR but not the overall SNR of the signal as assumed by STI. Our results in the Om conditions (i.e., speech understanding decreased with acoustic degradation) also support this notion.

Further, the above-presented conclusion appeared to be indirectly supported by studies investigating noise reduction algorithms using spectral subtraction. Spectral subtraction is a noise reduction strategy in which a speech in noise signal is transformed to the frequency domain, the estimated noise spectrum is subtracted from the speech in noise signal, and the speech with reduced noise signal is then converted back to the time domain. If the noise reduction algorithm could accurately estimate the noise spectrum, the within-channel SNR is improved in the processed signal. Several research groups have implemented the spectral subtraction noise reduction algorithm and reported improved speech recognition scores for CI users (Weiss, 1993; Hochberg *et al.*, 1992; Goldsworthy, 2005).

## V. SUMMARY AND CONCLUSIONS

Directional microphones and AMC noise reduction algorithms are two noise reduction strategies commonly used to reduce noise interferences in hearing aids. Both of these strategies increased the temporal modulation of speech envelope in background noise. The goals of this study were to determine whether directional microphones and AMC noise reduction algorithm could enhance speech recognition and/or sound quality in background noise, and to examine if a speech-based speech transmitted index could predict the speech intelligibility of CI users and normal hearing individuals listening to CI simulated speech. The long-term goal of this study was to investigate suitable signal processing strategies for enhancing CI performance.

The results of this study are encouraging: the directional microphone significantly enhanced speech recognition ability and overall preferences of CI users in background noise. Although the AMC noise reduction algorithm did not significantly improve speech recognition, it significantly improved cochlear implant users' sound quality ratings. Taken together, the rankings of speech recognition are Om=ON<Dm=DN and the rankings of overall preferences are Om<ON=Dm <DN. Overall, DN is the most desirable and Om is the least.

The positive findings in this study suggest that other advanced hearing aid technologies may also be implemented as preprocessors to CI speech processors to enhance CI performance and user convenience. Some examples include the microphone matching algorithm that can maintain directional effects of directional microphones over time, the switchless telecoil that can sense the magnetic field emitted by telephone headsets and automatically switch to telecoil input, before switching back to microphone input if the telephone headset is removed from the ear.

Previous studies reported that directional microphone performance may be reduced in real-life environments with reverberations (Hawkins and Yacullo, 1984; Ricketts and Hornsby, 2003). Some hearing aid studies reported that some directional hearing aid users obtained significant improvement in laboratory testing environments but they did not notice significant benefit in everyday life environments (Cord *et al.*, 2002; Mueller *et al.*, 1983; Ricketts and Hornsby, 2003; Surr *et al.*, 2002; Walden *et al.*, 2000). As the recordings of the present study were made in an anechoic chamber (i.e., little reverberation), field trials should be conducted to further evaluate the efficacy of directional microphones for CI users. In addition, as noise reduction algorithms rely on the differences in physical characteristics of speech and noise to separate speech and noise, the AMC noise reduction algorithm may not be effective if the background noise is speech. Further improvements are still needed to optimize the noise reduction algorithms to operate effectively in more acoustically complex environments.

Although STI used a mechanism similar to CI speech coding strategies to predict speech understanding, it successfully predicted the effects of noise and directional microphones but failed to predict the speech recognition scores of CI users for the AMC noise reduction algorithm used in this study. The results of this study also suggested that the within-channel SNR, instead of the overall level of noise, is the determining factor for speech understanding for CI users.

A caution in interpreting the results of this study is that the within-channel SNR in the CI speech processor depends on the number of signal processing channels in the AMC noise reduction algorithm and the CI speech processor. If an AMC noise reduction algorithm divided and processed signals in, say, 6 channels and then the processed signal is sent to a CI speech processor with 6 channels, the within-channel SNR in frequency channels with speech components may not

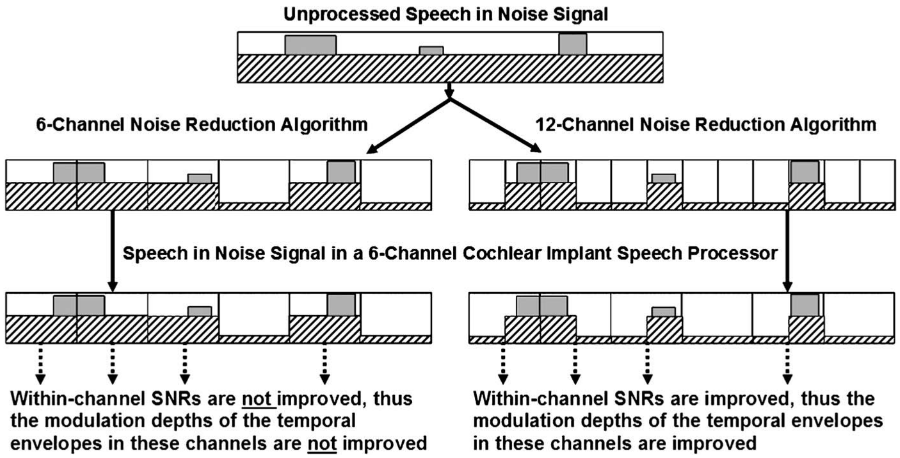


FIG. 8. The effects of the relative number of channels in the noise reduction algorithm and cochlear implant speech processor. Assume that this particular noise reduction algorithm does not reduce the gain of a frequency channel if speech, regardless of SNR, is detected in the channel. If the noise reduction algorithm has the same or lower number of channels than the cochlear implant speech processor (the example on the left side), the overall modulation of the temporal envelope is increased but the within-channel SNRs are not. On the other hand, if the noise reduction algorithm has higher number of channels than the cochlear implant speech processor (the example on the right side), both overall modulation of the temporal envelope and within-channel SNRs are increased.

be improved for the CI user because there are more channels in the speech processor than the noise reduction algorithm (Fig. 8). The reverse (i.e., the AMC noise reduction algorithm has 12 channels and the speech processor has 6 channels), however, could provide an increase in within-channel SNR because the noise reduction algorithm has a finer processing scale (Fig. 8). By the same logic, the within-channel SNR in the CI speech processor also depends on the relative spectrum of speech and noise as well as the cut-off frequencies of the frequency channels in the noise reduction algorithm and the speech processor. Therefore, the results of this study do not rule out the possibility that an AMC noise reduction algorithm with narrower frequency bands or more channels may enhance speech understanding of CI users with speech processor with fewer processing channels.

Further, if the above-mentioned assumptions hold, they support the implementation of noise reduction algorithms with more signal processing channels—as many channels as the signal processing power of the digital platform permits without significant overall processing delay—than CI speech processors. As implemented in the present form, it is possible that a noise reduction algorithm with higher number of channels can improve the within-channel SNR of a CI speech processor. Future studies are needed to determine if the effectiveness of the AMC noise reduction algorithm varies for different CI speech coding/electrical stimulation strategies and if noise reduction algorithms with a different number of signal processing channels would be more effective for CI users.

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