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Evaluation of different signal processing options in unilateral and bilateral Cochlear Freedom recipients using R-Space background noise

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EVALUATION OF DIFFERENT SIGNAL PROCESSING OPTIONS IN UNILATERAL AND BILATERAL COCHLEAR FREEDOM IMPLANT RECIPIENTS USING R-SPACE BACKGROUND NOISE

by

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Abstract: Understanding speech in the presence of noise is a difficult task for cochlear implant patients. This study examines the real-world effectiveness of different signal processing approaches available in the Cochlear Nucleus Freedom device to enhance speech perception in noise for cochlear implant recipients.

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INTRODUCTION AND REVIEW OF THE LITERATURE

The ability of cochlear implants to improve a patient's speech recognition has been well documented (Tyler & Moore, 1992; Skinner, Holden, Holden, Demorest, & Fourakis, 1997; Fetterman & Domico, 2002; Spahr & Dorman, 2004). There has been a dramatic improvement in speech recognition as cochlear implant (CI) speech processing strategies and equipment have advanced over the years. Spahr and Dorman (2004) reported that the average CI user scored 58% on word recognition tasks, 97% on sentence recognition tasks using clear speech, and 70% on sentence recognition tasks using conversational speech. The speech recognition in quiet was considerably better than the speech recognition in noise. When sentences were presented at a +10 decibel (dB) signal-to-noise ratio (SNR), speech recognition performance of average CI recipients decreased to 70% on tasks using clear speech and 42% on tasks using conversational speech. When presented at a +5 dB SNR, recognition of conversational speech sentences fell to 27%. Difficulty understanding in background noise is a common report among CI recipients.

The difficulty recognizing speech in noise for CI users is partially due to the loss of spectral resolution. Fu and colleagues (1998) demonstrated that fine spectral information may be critical for speech recognition in background noise and concluded that increasing the number of spectral channels for CI users will improve speech recognition in noise. However, there is a limit to the possible number of channels in the electrode array due to channel interaction and current spread. Friesen and colleagues (2001) hypothesized that most CI users cannot fully use the spectral information provided by the number of electrodes in their implant. The authors showed that the speech recognition in noise of the better performing CI users improved as the number of electrodes was increased to seven. The speech recognition in noise of the poorer performing CI users only improved with the number of electrodes increasing to four.

The difficulty recognizing speech in the presence of noise for CI recipients can also be partly due to the lack of binaural hearing. Improved speech recognition in noise is one of the benefits of binaural hearing (MacKeith & Coles, 1971; Gantz et al, 2002; Tyler et al, 2002; Tyler, Dunn, Witt, & Preece, 2003; van Hoesel & Tyler, 2003; Hawley, Litovsky, & Culling, 2004; van Hoesel, 2004; Brown & Balkany, 2007; Ching, van Wanrooy, & Dillon, 2007; Chan, Freed, Vermiglio, & Soli, 2008). Normal hearing individuals demonstrate this binaural benefit in noise and bilateral amplification helps preserve this benefit in hearing-impaired individuals (van Hoesel, 2004). Due to financial constraints and surgical risks, unilateral cochlear implantation is the standard clinical practice, but the occurrence of bilateral implantation is increasing as research continues to provide evidence of binaural benefit in bilateral CI recipients.

Improved speech recognition in noise with binaural hearing is thought to emerge from the combination of the head-shadow effect, binaural squelch, and binaural redundancy. The head-shadow effect occurs when speech and noise are spatially separated. For example, noise coming from the right side interferes with the signal in the right ear, but the head physically blocks some of the noise from reaching the left ear. Therefore, a better SNR is obtained at the left ear compared to the right, and the individual is able to selectively focus on the left ear to improve speech recognition (Tyler et al, 2003; Brown & Balkany, 2007; Ching et al, 2007). The head-shadow effect is frequency dependent because high-frequency sounds with wavelengths smaller than the size of the head are more easily blocked. High-frequency attenuation can be as much as 20 dB and low frequency attenuation is approximately 3 to 6 dB in the normal binaural auditory system (Feddersen, Sandel, Teas, & Jeffress, 1957; Tyler et al, 2003).

The binaural squelch effect also occurs when speech and noise are spatially separated so that the two ears receive different inputs. Central auditory processing analyzes the timing, amplitude, and spectral differences between the ears. This analysis creates a better representation of noise and speech helping the brain to effectively separate these signals. Even if the SNRs at the two ears are the same, the binaural squelch effect uses other differences between inputs to separate the speech and noise (Tyler et al, 2002; Tyler et al, 2003; Ching et al, 2007; Brown & Balkany, 2007). In addition, the binaural squelch effect uses localization cues to separate speech and noise (Tyler et al, 2002; Tyler et al, 2003). Binaural squelch provides an average improvement of approximately 2 dB (Ching et al, 2007).

Binaural redundancy occurs when identical signals are presented to both ears. The repetitive information to the brain from the separate auditory pathways enhances speech understanding in quiet and in noise. In noisy environments, binaural redundancy helps the brain form an accurate representation of the speech signal, in order to better separate it from the noise. The repeated message to the brain provides an increase in SNR of approximately 1 to 2 dB (Dillon, 2001; Ching et al, 2007).

Recent research has focused on measuring the effects of the head-shadow effect, binaural squelch, and binaural redundancy in bilateral CI recipients. Gantz and colleagues (2002) presented sentences from 0° azimuth at 70 dB SPL and multi-talker babble from either 90° or 270° azimuth. The SNR varied between patients to avoid floor and ceiling effects. Eight of the ten bilateral CI subjects showed significant improvement in speech recognition in noise with the addition of a second implant on the side away from the noise source, indicating a significant head-shadow effect. Four of the ten subjects demonstrated a significant improvement in speech recognition with the addition of the second implant on the side towards the noise, which is attributed to binaural squelch. The prevalence of binaural redundancy was also explored by presenting sentences and noise together from the front speaker. Sentences were again presented at 70 dB SPL with multi-talker babble presented at a +10 dB SNR. Four of the ten bilateral CI subjects exhibited a significant improvement in speech recognition with the addition of the second implant.

Tyler and colleagues (2002) used a configuration similar to the Gantz et al (2002) study, where sentences were presented at 70 dB SPL from 0° azimuth and noise was presented from either 90° or 270° azimuth. The level of the noise varied between subjects to control for floor and ceiling effects. All seven bilateral CI subjects showed a significant improvement in speech recognition with the addition of the second implant on the side away from the noise and three of the seven subjects showed significant improvement with the addition of the second implant on the side towards the noise source. Finally, when sentences and noise were presented together from the front speaker with sentences presented at 70 dB SPL in a +10 dB SNR, four of nine bilateral CI subjects showed significant improvement in speech recognition in noise. Tyler and colleagues (2002) observed a significant head-shadow effect, but similar to the findings of Gantz and colleagues (2002), the effects of binaural squelch and redundancy remain unclear.

Van Hoesel and Tyler (2003) presented sentences at 65 dB SPL from 0° azimuth and spectrally-matched noise from either 90° or 270° azimuths. An adaptive procedure was used to track the SNR corresponding to 50% correct. A considerable head-shadow effect was observed when the second implant was added to the side away from the noise. The authors attributed a 5 dB improvement in SNR to the head-shadow effect. Van Hoesel and Tyler (2003) also reported that binaural performance when the sentences and noise were both presented from the front was comparable to the better ear alone. Therefore, no effect of binaural redundancy was seen in any subjects tested.

Results from the three studies previously discussed all showed a significant head-shadow effect when a second implant was added to the side away from the noise source. Subjects were able to attend to the ear with the better SNR and thus speech recognition in noise improved. However, clear effects of binaural squelch and binaural redundancy are not evident in the research. The Tyler et al (2002) and Gantz et al (2002) studies reported significant effects of binaural squelch and binaural redundancy in less than half of their subjects, and van Hoesel and Tyler (2003) saw no effect of binaural redundancy. These findings suggest that the binaural advantages of squelch and redundancy can be beneficial to bilateral CI recipients, but they require the brain to successfully integrate information from both ears. A benefit from binaural squelch and redundancy may require more bilateral listening experience. It is also important to note that no deterioration in performance was seen when the second implant was added.

Whether the cochlear implant recipient has unilateral or bilateral CIs, understanding speech in the presence of background noise is one of the most challenging tasks. In order to improve speech recognition in noise, Cochlear Americas, the manufacturer of the Nucleus Cochlear Implant Systems, have incorporated a traditional dual-port directional microphone into their speech processors for many years. In this microphone arrangement, sound from behind reaches the rear port before the front port, creating an external time delay. The external time delay depends on the distance between the two microphone ports, which is seven millimeters in the Nucleus devices. The rear port uses an acoustic damper to create a low-pass filter. Sound entering the rear port is processed through the low-pass filter, producing an internal time delay. If the internal and external time delays are equal, sound from the rear will reach both sides of the microphone diaphragm at the same time, generating no net force and suppressing sounds from the rear direction. The direction of maximum suppression varies with the difference between the internal and external time delays (Dillon, 2001; Thompson, 2002).

Several speech processing options have been developed to improve speech recognition in noise while providing listening comfort. Options available in the Nucleus Freedom processor include Adaptive Dynamic Range Optimization (ADRO), Auto-Sensitivity Control (ASC), and BEAM. ADRO is a preprocessing strategy that repeatedly alters the gain of the input signal to place the signal optimally in the CI user's dynamic range. Gain is adjusted individually in each frequency channel according to a specific set of rules, which keeps the output level between a comfort target and audibility target (James et al, 2002; Dawson, Decker, & Psarros, 2004). Gain is increased if a sound falls below the audibility target or decreased if a sound rises above the comfort target. When the sound is within the audible and comfortable range, the gain operates in a linear fashion (Blamey, 2005). However, gain cannot exceed a specified maximum amount. This maximum gain rule works to limit the amplification of low-level background noise. Patients should hear low-level sounds, but these sounds should not be bothersome (James et al, 2002; Dawson et al, 2004). The rules enact slow-acting adjustments in channel gains using percentile estimates of the long-term output level of each frequency channel (Patrick, Busby, & Gibson, 2006).

ADRO was incorporated into the Nucleus CI system in 2002 as an input signal processing option (Patrick et al, 2006). Two studies have been conducted with CI recipients to determine the functional benefit of ADRO. James and colleagues (2002) compared the speech recognition of adult CI users in quiet and noise using ADRO and a standard speech processor program or map. ADRO demonstrated significantly better speech recognition scores on a closed-set spondee task presented at 40 dB SPL, a monosyllabic word recognition task presented at 60 dB SPL, and a sentence recognition task presented at 50 and 60 dB SPL. When sentences were presented at 70 dB SPL with eight-talker babble adjusted to +15 and +10 dB SNR, there was no significant difference in speech recognition between ADRO and the standard maps. Dawson and colleagues (2004) compared the performance of pediatric CI users on speech recognition tasks using ADRO and a standard map. ADRO showed a significant improvement in speech recognition when sentences were presented at 50 dB SPL in quiet and at 65 dB SPL in eight-talker babble with SNRs individually selected between 0 and +15 dB. The gain adjustments of ADRO lead to improved speech recognition at low and medium presentation levels; however, the ability of ADRO to improve speech recognition in noise is unclear.

Speech recognition in noise can also be improved by reducing the CI microphone sensitivity. Cochlear speech processors are equipped with a manual sensitivity control, which manipulates the automatic gain control (AGC) kneepoint. The AGC kneepoint is the input level at which compression begins. Below the kneepoint, amplification is typically linear (Dillon, 2001; Agnew, 2002b). When the sensitivity of the speech processor is reduced, the AGC kneepoint increases and when the sensitivity is increased, the AGC kneepoint decreases. Therefore, higher sensitivity (lower kneepoint) leads to more gain for soft sounds and greater audibility (Patrick et al, 2006).

CI users reported reducing the sensitivity in noisy environments to decrease the amplification of low-level background noise. This report led to the creation of autosensitivity control, or ASC. ASC is an optional preprocessing scheme that automatically adjusts the sensitivity according to the noise floor, or the intensity level of sound during breaks in speech. When the noise floor reaches the autosensitivity breakpoint, sensitivity is automatically decreased (kneepoint increased) to provide less low-level gain. When the noise floor falls below the breakpoint, sensitivity is automatically increased (kneepoint decreased) to provide more gain for soft sounds. At default settings, the autosensitivity breakpoint is 57 dB, and ASC aims to keep the noise floor at least 15 dB below the AGC kneepoint. The breakpoint can be changed in the software to make ASC more or less responsive to background noise. With ASC active, CI users typically perceive a decrease in the loudness of background noise (Patrick et al, 2006).

A new input signal processing scheme, BEAM, was introduced in the Nucleus Freedom speech processor in 2005. BEAM is a two-stage adaptive beamformer intended to improve the SNR. Figure 1 is a visual schematic showing the two stages (Wouters & Vanden Berghe, 2001). The first stage utilizes spatial preprocessing through a single-channel, adaptive dual-microphone system that combines the front directional microphone and rear omnidirectional microphone to separate speech from noise. The output from the rear omnidirectional microphone is filtered through a fixed finite impulse response (FIR) filter, a type of digital filtering characterized by a linear phase response (Agnew, 2002a). The output of the FIR filter is subtracted from an electronically delayed version of the output from the front, directional microphone to create the noise reference (Vanden Berghe & Wouters, 1998; Wouters & Vanden Berghe, 2001; Wouters, Vanden Berghe, & Maj, 2002; Spriet et al, 2007). The filtered signal from the omnidirectional microphone is then added to the delayed signal from the directional microphone to create the speech reference. This spatial preprocessing increases sensitivity for sounds arriving from the front while suppressing sounds that arrive between 90° and 270° azimuths. Directional polar plots comparing BEAM to the standard directional microphone are shown in Figure 2 (Patrick et al, 2006). The polar plot of the traditional directional microphone, shown in red, remains the same as the noise source moves. Maximum suppression is seen with the noise source located at 180° azimuth. The BEAM polar plots, shown in blue, adapt between cardioid, hypercardioid, and bidirectional patterns as the noise source moves to adjust the null points for maximum noise suppression. The second stage of BEAM utilizes adaptive noise cancellation to reduce the remaining noise in the speech reference. The filter coefficients used in the adaptive noise cancellation can only be adjusted during breaks in speech requiring a voice activity detector. These coefficients are then used to filter out the remaining noise in the speech reference (Wouters et al, 2002).

Wouters and Vanden Berghe (2001) investigated the speech recognition of four CI users utilizing a two-stage adaptive beamformer algorithm identical to the one used in BEAM processing. Monosyllabic words and numbers were presented at 0° azimuth at 55, 60, and 65 dB SPL in quiet and noise with the beamformer inactive and active. Speech-weighted noise was presented at a constant level of 60 dB SPL from a source located at 90° azimuth on the side with the implant. Word recognition in noise was significantly better for all presentation levels with the beamformer active, showing an average SNR improvement of more than 10 dB. Number recognition in noise was also significantly better with the beamformer active, demonstrating an average SNR improvement of 7.2 dB across conditions. Wouters and Vanden Berghe (2001) concluded that the two-stage adaptive beamformer lead to significant improvement in speech recognition in noise for CI users.

Spriet and colleagues (2007) investigated the performance of the BEAM preprocessing strategy in the Nucleus Freedom speech processor with five CI users. Subjects repeated sentences presented from a speaker at 0° azimuth in the presence of different types, levels, and locations of background noise using the standard directional microphone and BEAM. Speech-weighted noise and multi-talker babble were presented at constant levels of 55 and 65 dB SPL from either one source located at 90° azimuth or from three sources located at 90°, 180°, and

270° azimuths. The SNR was calculated using an adaptive procedure where the sentence presentation level was adjusted depending on correct or incorrect responses. BEAM improved the average SNR in all conditions when compared to the standard directional microphone. Improvement ranged from 1.5 dB with 55 dB SPL speech-weighted noise presented from three locations to 15.9 dB with 65 dB SPL multi-talker babble presented from one location. Noise was presented at two intensity levels, 55 and 65 dB SPL. BEAM processing resulted in greater mean SNR improvement at the louder noise level of 65 dB SPL, indicating that BEAM processing was more beneficial in louder environments.

The Spriet et al (2007) study also measured the percent phonemes correct for monosyllabic words presented in noise. Words were presented at constant levels of 55 and 65 dB SPL from 0° azimuth. The noise level was held constant at 60 dB SPL, but the type and location of the noise varied as previously discussed. Again, BEAM showed an increase in percent phonemes correct on the word recognition task for all conditions when compared to the standard directional microphone. Improvement ranged from 3% with words presented at 65 dB SPL in the presence of babble from three sources to 41% with words presented at 55 dB SPL in the presence of speech-weighted noise from one source. Spriet and colleagues (2007), similar to the Wouters and Vanden Berghe (2001) study, concluded that the available two-stage, adaptive beamformer, BEAM, enhances the SNR and leads to improved speech recognition in background noise.

Processing strategies similar to those found in the Freedom device, including traditional directional microphones and adaptive beamformers are found in hearing aids, and their effectiveness has been demonstrated in traditional laboratory simulations (Valente, Fabry, & Potts, 1995; Valente, Schuchman, Potts, & Beck, 2000; Pumford, Seewald, Scollie, & Jenstad, 2000). The benefit of these strategies in hearing aids ranges from about a 5.5 to 11 dB improvement in SNR or 40 to 70% improvement on speech recognition tasks in noise (Ricketts & Dittberner, 2002). The large variability in measured benefit depends on the specific processing strategy tested and the level, type, and location of noise used in testing. However, the improvement measured in the laboratory is often not considered representative of improvement found in real-world situations. The difficulty of effectively evaluating an individual's performance in a way that reflects real-life listening situations is an often recognized concern in hearing research. To address this issue, Compton-Conley and colleagues (2004) developed an eight loudspeaker test system to replicate a restaurant environment, the R-Space.

A study was conducted by the developers to assess the accuracy of the R-Space and other typical measures of directionality in determining directional microphone benefit in hearing aids. Three methods of simulating restaurant noise were employed: the R-Space, noise from a single source behind the listener, and noise from a single source above the listener. These simulations were then compared to measurements taken at an actual restaurant, the live condition. Results of the study showed that only the R-Space simulations provided accurate estimates of the absolute performance of directional microphones. Performance in the R-Space condition was not significantly different from performance in the live condition (Compton-Conley, Neuman, Killion, & Levitt, 2004).

The current data available on the benefit of different CI processing strategies in noise were obtained using traditional laboratory measures and may not be accurate predictors of real-world performance. Since the Freedom processor uses strategies similar to those found in hearing aids, the Compton-Conley et al (2004) study supports the use of the R-Space to better predict the real-world effectiveness of the different signal processing approaches available to enhance speech perception in noise for CI recipients.

The goal of the present study was to measure speech recognition of CI recipients in background noise with the R-Space. Four signal processing options were measured that include standard directional, ADRO, ASC, and BEAM at two different noise levels, a moderate intensity level of 60 dB SPL and a loud intensity level of 70 dB SPL. It was hypothesized that speech recognition in noise would be greatest when the BEAM processing was utilized and least with the standard directional processing. It was also hypothesized that BEAM would show greater mean SNR improvement at the louder noise level. This study may help determine the speech processing option that yields better speech recognition in background noise for CI recipients, resulting in increased patient benefit and satisfaction.

METHODS

Subjects

Thirty subjects, twenty-seven unilateral and three bilateral CI recipients, participated in this study with a mean age of 60 years (range of 25-82 years; SD=15.2). Thirteen of the participants were female and 17 male. For the bilateral subjects, the data from one ear was randomly selected and included in the unilateral data analysis. Varying etiologies and length of hearing loss and hearing aid use prior to implantation were evident in this sample. The right ear was implanted in 14 subjects and the left ear was implanted in 16 subjects. The mean years of hearing loss and years of severe-to-profound hearing loss prior to implantation was 30.7 (range of 1-54 years; SD=13.9) and 13.8 (range of 1-45 years; SD=13.8), respectively. The mean years of hearing aid use prior to implantation in this sample was 20.3 (range of 0-47 years; SD=14.2). Table 1 reports individual demographic and hearing history information for unilateral subjects.

All subjects were implanted with the Nucleus 24 Contour or Contour Advance internal array and had maps programmed following a clinical protocol developed at Washington University School of Medicine (Skinner, 2003). The mean years of implant use for unilateral subjects was 3.4 (range of 0.5-7.9 years; SD=2.0). Twenty-seven of the thirty subjects used the Advanced Combination Encoder (ACE) strategy. The remaining three subjects used Spectral Peak (SPEAK), Continuous Interleaved Sampling (CIS), and MP3000, a research strategy previously studied at the Washington University Adult Cochlear Implant Center. All subjects were required to have open-set speech recognition to be included in the study. CNC monosyllabic word scores in quiet ranged from 17 to 86% with a mean score of 56.8% (SD=20%). Specific programming information is reported in Table 2.

Bilateral subjects (numbers 2, 8, and 9) had a mean of 3.3 years (range of 2-4.5 years; SD=1.3) between the first and second implant and a mean of 2.7 years (range 1.7-3.4 years; SD=0.9) of bilateral use at the time of testing. Table 3 reports the hearing history and specific programming information of the bilateral subjects for the ear not included in the unilateral analysis.

Approval for this study (#08-1038) was obtained from the Washington University School of Medicine Human Research Protection Office (HRPO) prior to data collection. Subjects signed an informed consent document approved by the HRPO committee, which outlined the testing procedures and the risks and benefits of this study. Subjects were reimbursed $10 for the time spent testing, $0.24 per mile for travel, and $2.00 for parking.

Equipment/Test Environment

The Nucleus 24 Contour and Contour Advance internal arrays used in this study consist of a receiver/stimulator with 24 electrodes, 22 intracochlear electrodes and two extracochlear electrodes (Parkinson et al, 2002). The Nucleus Freedom Processor houses the microphones and the main computer, which controls sound processing. The microphones pick up external sound, which is then converted to a digital signal and sent to the internal implant. A Microsoft Windows programming system (Custom Sound version 2.0 developed by Cochlear Americas) is used to program the speech processor. The speech processor is hardwired to a programming interface (Processor Control Interface) connected to a personal Dell computer equipped with the Custom Sound 2.0 software. The speech processing strategies implemented by this system include Spectral Peak (SPEAK), Advanced Combination Encoder (ACE), and Continuous Interleaved Sampling (CIS) (Skinner, Arndt, and Staller, 2002). A Nucleus Freedom processor was obtained from Cochlear Americas, and all unilateral subjects were tested using the same processor to ensure the equipment was performing optimally. Bilateral subjects were tested using the processor from Cochlear Americas and an available loaner processor from the Washington University School of Medicine Adult Cochlear Implant Center.

Eight loudspeakers were positioned in a 360° arc, with loudspeakers spaced in increments of 45° around the listener. The subject was seated in the center of the arc, 24 inches from each loudspeaker. Each loudspeaker is 44 inches above the ground at ear level for a seated average height adult (see Figure 3). All testing was completed in a double-walled sound-treated booth (8’3’’x 8’11’’).

A Dell personal computer with a sound card, a power amplifier (Crown, Model D-150), and a custom designed mixing and amplifying network (Tucker-Davis Technologies) was utilized for presenting warble tones in the soundfield to measure aided thresholds.

An Apple IMAC 17 personal computer with a 2 GHz Intel Core 2 Duo Processor, 2 GB of memory, and MAC OS 10 operating system was used to operate the R-Space, a speech in real-world environmental noise reproduction system. The R-Space configuration was implemented via professional audio mixing software (MOTU Digital Performer 5) and an audio interface (MOTU 828mkII, 96 kHz firewire interface). The output of the audio interface was sent to four amplifiers (ART SLA-1, two-channel stereo linear power amp with 100 watts per channel) and then to the eight loudspeakers (Boston Acoustic CR67) set in a circular array.

Test Materials

The Hearing in Noise Test (HINT sentences) consists of 25 recorded, phonetically balanced lists of 10 sentences each. The lists were recorded by a male talker of American English and are intended for adaptive measurement of Reception Threshold for Sentences (RTS) in quiet or in noise (Nilsson, Soli, and Sullivan, 1994).

The R-Space noise is a live recording made at a restaurant using an eight microphone array, developed specifically for use in the R-Space. To record the noise, the Knowles Electronic Manikin for Acoustic Research (KEMAR) was equipped with a circular, horizontal array of eight interference-tube microphones placed in equal 45° increments around his head. The eight microphone set-up allowed the noise surrounding KEMAR to be recorded as it reached the head. The recording was made at a busy neighborhood restaurant (Compton-Conley et al, 2004). The recording consists of real-life uncorrelated noise that occurs at a restaurant, including sounds of dishes clanking, people talking, and background music.

Frequency-modulated (FM) tones (centered at 250, 500, 1000, 2000, 3000, 4000, and 6000 Hz), sinusoidal carriers modulated with a triangular function over standard bandwidths recommended for use in the soundfield by Walker, Dillon, and Byrne (1984), were used to obtain aided soundfield thresholds prior to testing.

Calibration

For calibration of HINT sentences and the R-Space noise, the sound level meter (Bruel & Kjaer, Model 2230) was placed with the microphone (Bruel & Kjaer, Model 4155) at 90° azimuth to the stimulus in the center of the R-Space array at the height of the center of the loudspeakers. Measurements were made with 0 dB attenuation using a dB sound pressure level (SPL) linear –shaped scale. For the HINT sentences, the overall SPL of all lists was taken as the average of the peaks on the slow, RMS, linear scale through the front loudspeaker. The maximum output was recorded as 83.7 dB SPL. For the R-Space noise, an equivalent continuous sound pressure level measure was obtained for five minutes with the sound level meter set using dB L_{eq}. The maximum output was 73.9 dB SPL. The magnitude of attenuation was chosen based on the measured maximum output and the desired intensity level of the signal.

Test Procedures

Aided Soundfield Thresholds

FM tone soundfield thresholds were obtained at 250, 500, 1000, 2000, 3000, 4000, and 6000 Hz in a modified Hughson-Westlake procedure (Carhart and Jerger, 1959) with a +2 and -4 dB HL step size. Mean soundfield thresholds were obtained in the standard directional map to verify audibility with the speech processor. Mean soundfield thresholds are shown in Figure 4.

Reception Threshold for Sentences

Two lists of HINT sentences were presented from the loudspeaker located at 0° azimuth with the R-Space noise presented from all eight loudspeakers at two constant noise levels, 60 and 70 dB SPL. A Reception Threshold for Sentences (RTS) was obtained using an adaptive procedure in 2 dB step sizes. The level of sentence presentation was adjusted based on correct or incorrect response. If a correct response was obtained, the presentation level of sentences decreased 2 dB. If an incorrect response was obtained, the presentation level of sentences increased 2 dB. RTS is equivalent to SNR and was calculated by averaging across sentences and subtracting the noise level. One practice list was presented to familiarize the subjects with the task.

The four listening conditions, standard (STD), ADRO, ASC, and BEAM, were tested at two noise levels, 60 and 70 dB SPL. The subject's preferred settings (map) with no additional processing was used for the STD condition. The processing options (ADRO, ASC, and BEAM) were added to the STD map. The subject's everyday volume and sensitivity settings were used for all conditions. The non-test ear was plugged when hearing thresholds were 60 dB HL or better. Processing options (STD, ADRO, ASC, and BEAM), intensity levels (60 and 70 dB SPL), and lists were randomly assigned. For unilateral CI subjects, all testing was performed in one session. Bilateral implant subjects attended two sessions, one for each ear, with the bilateral condition tested at 60 dB SPL in the first session and 70 dB SPL in the second.

Statistical Analysis

Unpaired t-tests were performed to compare RTSs within processing options and noise levels, and a mixed model repeated measures analyses of variance (ANOVA) was used to analyze RTSs across all combinations of processing options and noise levels. An unstructured covariance structure was designated within the mixed model to account for the completely within subject crossed study design with a focus on the noise level x processing option interaction. This interaction tested the hypotheses regarding the equality of changes across noise levels and processing options. Tukey-adjusted p-values within the ANOVA model were used to determine significance ($p \leq 0.05$) for pairwise comparisons.

Demographic and audiologic variables were also investigated to determine if any impacted the interaction between noise level and processing option treatment levels. The variables of interest included the implanted ear, subject age at testing, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation, length of implant experience, and speech recognition scores in quiet on a CNC word task. The three-way interaction between these potential moderating variables, processing option, and noise level could not be explored due to sample size limitations. As a result, the potential moderating variables were divided into groups. Continuous variables were divided into two categories using the group median value. Implanted ear was already inherently dichotomized. Unpaired t-tests were used to compare data between potential moderating variable groups within processing options and noise levels, and a mixed model ANOVA was used to explore the noise level x processing interaction within potential moderating variable groups. If no significant interaction was found, the interaction was dropped from the mixed model and the main effects of processing option and noise level were investigated. All data analysis was produced using SAS software, version 9.2 of the SAS System for Linux (SAS Institute Inc., Cary, NC, USA).

RESULTS

Unilateral Subjects

Statistical analyses identified both noise level [$F(1,29)=29.8$; $p<0.0001$] and processing option [$F(3,29)=22.3$; $p<0.0001$] as significant main effects. A significant [$F(3,29)=5.18$; $p=.006$] noise level x processing option interaction was also identified, indicating that processing is differentially affected by noise level. The four processing options investigated showed different patterns of change with increasing noise level. Due to the significant interaction, the effect of noise level and processing option independent of each other was not meaningful.

In 60 dB SPL noise with STD processing, subjects exhibited a mean RTS of 10.8 dB (SD=6.8). ADRO processing had the poorest performance with a mean RTS of 12.8 dB (SD=7.4). ASC and BEAM processing showed an improvement in RTS relative to STD and ADRO processing with means of 9.5 (SD=6.8) and 8.3 (SD=7.1) dB, respectively (see Figure 5). BEAM processing resulted in a statistically significant improvement in the mean RTS relative to both STD [$t(29)=-3.82$; $p\leq 0.05$] and ADRO processing [$t(29)=5.13$; $p\leq 0.05$]. The mean RTSs for STD, ADRO, and ASC were not statistically different from each other, although ASC performed 1.3 and 3.3 dB better than STD and ADRO processing, respectively. There was also no statistical difference between ASC and BEAM processing.

In 70 dB SPL noise, STD and ADRO processing showed similar performance, with mean RTSs of 15.6 (SD=4.9) and 15.0 (SD=5.4) dB, respectively. ASC processing had significantly better mean RTSs compared to STD [$t(29)=-6.87$; $p\leq 0.05$] and ADRO processing [$t(29)=6.36$; $p\leq 0.05$]. BEAM processing also exhibited significantly better RTSs than STD [$t(29)=-5.29$; $p\leq 0.05$] and ADRO [$t(29)=4.87$; $p\leq 0.05$] processing. ASC processing had the best mean RTS of the four conditions (9.7 dB, SD=5.7), followed by BEAM processing with a mean RTS of 11.4 dB (SD=7.3). No significant differences were observed between STD and ADRO or between ASC and BEAM (see Figure 6).

The subjects' performance was poorer in all processing conditions when the noise level was increased from 60 to 70 dB SPL. The decrease in performance varied among processing conditions. The smallest change was for ASC processing, whose performance decreased by only 0.2 dB. STD processing had the largest change with a decrease in performance of 4.8 dB. ADRO exhibited a decrease in performance of 2.2 dB, and BEAM showed a decrease of 3.1 dB when the noise level increased. STD [$t(29)=-3.94$; $p\leq 0.05$] and BEAM [$t(29)=-5.16$; $p\leq 0.05$]

processing resulted in significantly poorer RTSs in 70 dB SPL noise compared to the performance with these processing conditions at 60 dB SPL. There was no statistical difference between noise levels for ADRO and ASC processing (see Figure 7).

<u>Moderating Variables</u>

When divided either categorically or by the median, all variables investigated were found to be significant moderators for the noise level x processing option interaction. To explore the effect of implanted ear, subjects were divided into right ear and left ear groups. The right ear and left ear group showed different associations between processing condition and noise level. The right ear group reported a significant noise level x processing interaction [$F(3,13)=3.82$; $p=0.04$]. The processing options revealed different patterns of change when the noise level increased from 60 to 70 dB SPL. STD processing demonstrated a significant ($p \leq 0.05$) decrease in performance of 6.5 dB with the increase in noise level. ADRO and BEAM processing reported decreases in performance of 3.2 and 3.0 dB, respectively, with the increase in noise, and ASC demonstrated a decrease in performance of only 0.5 dB. The left ear group revealed significant main effects of noise level [$F(1,15)=7.96$; $p=0.01$] and processing option [$F(3,15)=14.4$; $p=0.0001$] with no significant interaction. The left ear group showed essentially stable performance with ASC processing across noise levels, but performance decreased by as much as 3.4 dB for STD processing with the increase in noise level.

To investigate the effect of age at testing, subjects were divided into two groups by the median age of 61 years. Different effects on the association between processing condition and noise level were seen for the two groups. Younger subjects revealed a significant noise level x processing interaction [$F(3,14)=4.24$; $p=0.03$], indicating that the processing condition was differentially affected by noise level. STD processing performance decreased significantly

($p \leq 0.05$) by 5.0 dB when the noise level was increased from 60 to 70 dB SPL. ADRO and BEAM processing performance decreased 1.8 and 3.7 dB, respectively, with the increase in noise level, but ASC performance improved 0.4 dB with the increase in noise. Older subjects, however, demonstrated significant main effects of noise level [$F(1,14)=25.4$; $p=0.0002$] and processing condition [$F(3,14)=19.9$; $p<0.0001$] with no significant interaction. The older subjects performed poorer at 70 than at 60 dB SPL for all processing conditions. The average decrease in performance ranged from 0.8 dB with ASC processing to 4.6 dB with STD processing.

To investigate the effect of years of hearing loss, subjects were divided into two groups by the median value of 31 years. The association between noise level and processing condition was different for the two groups. Subjects with a history of hearing loss of more than 31 years prior to implantation revealed a significant noise level x processing interaction [$F(3,15)=6.24$; $p=0.006$], suggesting that the four processing conditions were affected differently by the increasing noise level. STD and BEAM processing performance decreased by 4.7 and 3.0 dB, respectively, with the increase in noise level. Performance with ADRO decreased only 0.8 dB, and ASC performance improved by 2.1 dB with the noise level increase from 60 to 70 dB SPL. Subjects with less than 31 years of hearing loss resulted in statistically significant main effects of noise level [$F(1,13)=105.7$; $p<0.0001$] and processing condition [$F(3,13)=37.6$; $p<0.0001$] with no significant interaction. These subjects performed poorer at 70 than at 60 dB SPL for all processing conditions, with average decreases in performance ranging from 2.8 dB with ASC processing to 4.9 dB with STD processing.

Subjects were divided into two groups by the median value of 7 years to investigate the effect of years of severe-to-profound hearing loss prior to implantation. Different effects on the association between noise level and processing condition were observed. The results of subjects with more than 7 years of severe-to-profound hearing loss demonstrated a significant noise level x processing interaction [$F(3,14)=4.53$; $p=0.02$]. Performance with STD and BEAM processing decreased by 4.4 and 3.2 dB, respectively, ADRO performance decreased by 1.6 dB, and performance with ASC processing improved by 1.5 dB with the increase in noise level. However, the subjects with less than 7 years of severe-to-profound hearing loss at the time of implantation revealed significant main effects of noise level [$F(1,14)=64.8$; $p<0.0001$] and processing condition [$F(3,14)=25.8$; $p<0.0001$] with no significant interaction. These subjects performed poorer at 70 than at 60 dB SPL for all processing conditions, with average decreases in performance ranging from 1.1 dB with ASC processing to 5.2 dB with STD processing.

To explore years of hearing aid use prior to implantation, subjects were divided into two groups by the median value of 20 years. The performance of subjects with more than 20 years of hearing aid experience exhibited a significant noise level x processing interaction [$F(3,14)=4.56$; $p=0.02$], indicating that processing conditions were differentially affected by noise level. STD and BEAM processing performance decreased by 4.8 and 2.6 dB, respectively, ADRO performance decreased by 1.2 dB, and ASC processing performance improved by 1.9 dB with the noise level increase from 60 to 70 dB SPL. Subjects with less than 20 years of hearing aid experience demonstrated significant main effects of noise level [$F(1,14)=43.9$; $p<0.0001$] and processing condition [$F(3,14)=35.1$; $p<0.0001$] with no significant interaction. These subjects performed poorer at 70 than at 60 dB SPL for all processing conditions, with average decreases ranging from 2.3 dB with ASC processing to 4.8 dB with STD processing.

Years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation were highly correlated. Subjects with more years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid experience prior to implantation showed a significant interaction between noise level and processing condition, and the same patterns of change were observed. Subjects with less years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use showed no interaction, but significant main effects of noise level and processing condition were seen.

To investigate the effect of years of implant use, subjects were divided by the median value of 3.3 years. The results of subjects with more than 3.3 years of implant experience demonstrated a significant interaction between noise level and processing condition [$F(3,14)=8.99$; $p=0.001$]. Performance of the different processing conditions showed different patterns of change with increasing noise level. Performance with STD, ADRO, and BEAM processing worsened by 6.9, 3.6, and 3.7 dB, respectively, with the increase in noise, while ASC performance improved by 1.1 dB. Subjects with less than 3.3 years of implant experience showed a significant main effect of processing condition [$F(3,14)=10.9$; $p=0.0006$]. Noise level was not significant and no interaction was found. Therefore, ASC and BEAM processing performed better than STD and ADRO and noise level was irrelevant to the outcome.

To explore the effect of speech recognition in quiet, subjects were divided into two groups by the median CNC score of 57%. Within all processing conditions and noise levels, subjects with CNC scores above 57% performed significantly better ($p \leq 0.03$) than subjects with CNC scores less than 57%. The group with higher CNC scores revealed better performance across noise levels and processing conditions when compared to the group with lower speech recognition in quiet. Also, the group with higher CNC scores demonstrated a significant noise level x processing interaction [$F(3,15)=4.11$; $p=0.03$], indicating that processing conditions were differentially affected by noise level. STD and BEAM processing performance significantly

($p \leq 0.05$) decreased by 6.3 and 4.4 dB, respectively, and ADRO performance decreased by 3.0 dB, and ASC performance decreased by only 0.4 dB as the noise level increased. The group with lower CNC scores exhibited a significant main effect of processing condition [$F(3,13)=7.33$; $p=0.004$]. However, noise level was not significant. Performance for subjects with lower CNC scores was better with ASC and BEAM than STD and ADRO and noise level was irrelevant to the outcome.

Years of implant use and speech recognition scores in quiet were highly correlated ($r=0.48$; $p=0.008$) and similar results were found. Subjects with more years of implant experience and higher CNC scores showed a noise level x processing interaction, whereas subjects with less implant experience and lower CNC scores showed only a significant main effect of processing condition.

Bilateral Subjects

Due to the small sample size, no statistical analyses could be performed on the bilateral data, but performance for the three bilateral CI subjects is described. For the three bilateral CI subjects, bilateral improvement was evident for STD, ADRO, and BEAM processing with noise at 60 dB SPL. STD processing revealed a mean improvement of 1.4 dB when comparing the bilateral condition to the better monaural ear condition. ADRO had a mean bilateral improvement of 1.3 dB, and BEAM had a mean improvement of 3.0 dB. BEAM processing resulted in the best bilateral performance at 60 dB SPL with a mean RTS of 1.6 dB. Best performance with ASC processing was seen for the left ear alone condition. Mean bilateral performance with ASC processing decreased 2.1 dB compared to performance in the left ear alone condition.

All processing conditions showed a considerable bilateral improvement in 70 dB SPL noise. STD processing resulted in a bilateral improvement of 2.5 dB when compared to the better monaural ear condition. ADRO processing revealed a mean RTS bilateral improvement of 7.2 dB, and ASC bilateral performance improved 4.7 dB. The RTS improved the most between unilateral and bilateral conditions with BEAM processing with a mean improvement of 9.7 dB. In 70 dB SPL noise, BEAM processing had the best bilateral performance with a mean RTS of 0.4 dB.

Monaural performance for the bilateral subjects typically followed the trend of the unilateral subjects, showing a decrease in performance when noise level increased from 60 to 70 dB SPL. This trend in performance occurred for only one processing condition when the subjects were stimulated bilaterally. Bilateral performance with STD processing declined when the noise level increased from 60 to 70 dB SPL. The STD processing bilateral RTS was 3.7 dB poorer at the louder noise level. Bilateral performance with ADRO, ASC, and BEAM processing was different than the group monaural performance in that performance improved when increasing the noise level from 60 to 70 dB SPL. ADRO, ASC, and BEAM had mean RTS improvements of 3.3, 2.8, and 1.2 dB, respectively.

DISCUSSION

The results of this study suggest that on average ASC and BEAM processing improve speech recognition in noise for CI recipients in moderate and loud noise levels with ASC exhibiting best performance at the loud noise level and BEAM exhibiting best performance at the moderate noise level. This finding supports CI recipients' subjective reports of preferences for different processing options in different listening environments. ADRO processing on average demonstrated results similar to STD processing. STD and BEAM processing revealed a decrease in performance in the loud noise level compared to the performance of these processing conditions at the moderate level, while ADRO and ASC performance remained stable across noise levels.

Previous studies have investigated the ability of ADRO processing to improve speech recognition in noise for CI recipients. James and colleagues (2002) concluded that adult CI performance in noise with ADRO processing was not significantly different than performance with standard processing, but Dawson and colleagues (2004) observed a significant improvement in speech recognition in noise with ADRO processing compared to standard in pediatric CI recipients. The results of the current study support the findings of James et al (2002). No significant difference in speech recognition in noise for CI recipients was seen between mean performance with ADRO and STD processing. ADRO performance also remained stable when the noise level was increased. This stability across noise levels can probably be explained by the maximum gain rule of ADRO processing. The maximum gain rule states that gain cannot exceed a specified maximum amount. In 60 dB SPL noise, the amplification of background noise has already met the maximum amount of allowable gain and therefore, no additional amplification is provided when the noise level is increased.

The current results revealed that ASC processing can improve speech recognition in noise for CI recipients at both moderate and loud noise levels compared to STD processing with mean RTS improvements of 1.3 and 6.0 dB, respectively. ASC works to limit the processing of background noise by increasing the AGC kneepoint. Increasing the kneepoint limits the amplification of distant, softer sounds, and amplifies the closer, louder sounds. ASC operates on the assumption that the desired signal is closest to the subject. One would postulate that in the diffuse noise environment of the R-Space, where the noise sources and speech are the same distance from the subject that ASC processing would not aid in speech recognition. However, current results do not support this notion. Perhaps the hardware directional microphone provides sufficient directionality in conjunction with ASC processing to increase sensitivity of sounds arriving from the front and maximize suppression of background noise from other directions. Performance with ASC also remained stable across noise levels. This suggests that increasing the kneepoint as the noise floor increases is an adequate method to limit amplification of background noise.

The ability of BEAM to improve speech recognition in noise for CI recipients has been demonstrated in previous research. Wouters and Vanden Berghe (2001) reported an average improvement in SNR of greater than 10 dB on word recognition tasks with the beamformer algorithm, and Spriet and colleagues (2007) found improvement ranging between 1.5 and 15.9 dB. However, these models used laboratory simulations with speech-weighted noise and multi-talker babble presented from either one or three distinct noise sources. Current results show an RTS improvement of 2.5 dB at 60 dB SPL and 4.2 dB at 70 dB SPL, with noise presented from the diffuse field of the R-Space. These findings support the previous literature showing that BEAM can lead to a more desirable SNR and improve speech recognition in noise for CI recipients.

The current study, however, demonstrated less improvement in speech recognition in noise with BEAM processing than previous studies. These differences may be explained by the noise source used. The R-Space configuration presents noise from all eight loudspeakers. Therefore, the front speaker presents both speech and noise. BEAM utilizes directionality to divide speech from spatially separated noise. When the speech and noise are presented together from the front speaker, BEAM relies on the adaptive noise cancellation stage to reduce the noise.

BEAM may be more effective at improving speech recognition in noise when the noise source is spatially separated from the speech signal. Since typical real-world listening situations often include combined speech and noise, previous studies may have overestimated the absolute performance of BEAM, and current results may better predict the performance of BEAM processing in real-world situations similar to that replicated by the R-Space.

The differences in improvement in speech recognition in noise between the current study and previous research may also be explained by the noise type. Valente, Mispagel, Tchorz, and Fabry (2006) investigated the difference in RTS for bilateral hearing aid users when HINT sentences were presented in the R-Space configuration using HINT noise and R-Space noise. The RTS for R-Space noise was 1.3 dB poorer than the RTS for HINT noise. Therefore, speech recognition tasks may be more difficult when the R-Space noise is used compared to other continuous noise types, which might also contribute to the differences in RTS improvement between the present study and previous studies.

The results from the unilateral subjects also suggest that the variables of implanted ear, age, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation, length of implant experience, and speech recognition in quiet affect the association between processing and noise level. Therefore, these variables can be predictive of the benefit a patient will experience from these processing options with changing noise levels. For example, a patient with the right ear implanted may be able to better maintain performance when the noise level increases using ASC, but a patient with the left ear implanted may experience significant decreases in performance when the noise level increases for all processing options.

The current results also suggest that typical CI recipients will benefit from the use of ASC and BEAM processing in noisy listening environments. Spahr and Dorman (2004) reported that the average CI user scored 58% on a word recognition task in quiet. The average CNC word recognition score in quiet of the unilateral subjects in the present study was 56.8%. The comparable scores of speech recognition in quiet between the two studies suggest that the current sample of unilateral CI subjects is representative of the unilateral CI population.

Bilateral subjects demonstrated a bilateral benefit for all processing conditions, except ASC in 60 dB SPL noise. This supports the findings of previous studies showing improved speech recognition in noise with binaural hearing. Earlier studies (Gantz et al, 2002; Tyler et al, 2003; van Hoesel & Tyler, 2003) attribute the majority of binaural benefit to the head-shadow effect. However, in this study, equal SNRs would be present at the two ears due to the symmetric, diffuse noise source of the R-Space and the uncorrelated nature of the R-Space noise. Therefore, the bilateral improvement observed in this study is more likely attributed to binaural squelch and redundancy. Although the SNRs are equal at each ear, identical noise is not presented from each speaker, allowing the brain to use differences in the timing and spectrum of the input signal to separate the speech and noise. Also, the speech presented from the front loudspeaker is perceived by both ears providing redundant information to the brain. This redundancy allows the brain to develop a better representation of the message.

Previous studies (Gantz et al, 2002; Tyler et al, 2003; van Hoesel & Tyler, 2003) testing bilateral CI recipients' speech recognition in noise have shown unclear effects of binaural squelch and redundancy. The current results with three bilateral subjects show a mean bilateral improvement as high as 9 dB with the BEAM processing at 70 dB SPL. The variation in results between studies could be ascribed to characteristics of the individual subjects. For example, the

Tyler et al study (2002) measured speech recognition in noise with bilateral subjects after only three months of bilateral implant experience. The three subjects in this study have had a minimum of 1.7 years of bilateral implant experience. They also demonstrate similar speech understanding in quiet with each ear alone, which may allow better integration of the binaural signal in noise. Also, the difference in noise types and arrays may play a role in the variation. Previous studies have typically measured the effects of binaural squelch by presenting speech from 0° azimuth and noise from 90° azimuth. The monaural condition consists of the ear contralateral to the noise alone, and the binaural condition consists of adding the second ear with a poorer SNR. The R-Space presents noise to both ears at the same time, but the noise is uncorrelated. Therefore, the R-Space noise may better demonstrate the brain's ability to analyze the differences and similarities between inputs from the two ears to improve the internal representation of speech and noise. However, the small sample size of the current study makes comparisons to previous studies difficult.

A considerable bilateral advantage was evident in the three bilateral subjects of this study, and the degree of bilateral improvement increased as the noise level increased. Similar to the results from unilateral subjects, monaural performance of bilateral subjects was poorer at 70 than at 60 dB SPL, however, bilateral performance improved for ADRO, ASC, and BEAM processing conditions with increasing noise level. The increase in bilateral improvement found at the higher noise level suggests that bilateral benefit is greater as the listening situation becomes more challenging.

Although ASC and BEAM processing have been shown to improve the speech recognition in noise for unilateral and bilateral CI recipients, speech recognition in noise for CI recipients is still drastically poorer than that for normal hearing individuals. Nilsson, Gelnett, Sullivan, and Soli (1992) developed norms for the HINT using 150 normal hearing individuals. Sentences were presented from 0° azimuth with 65 dB A spectrally-matched noise presented from either 0°, 90°, or 270° azimuth. The average SNR needed to obtain 50% performance was found to be -2.7 dB. In this study, ASC and BEAM processing demonstrated mean RTSs ranging from 8.3 to 11.8 dB for unilateral subjects, more than 10 dB greater than that reported by Nilsson and colleagues for normal hearing individuals. Mean RTSs of ASC and BEAM processing for bilateral subjects ranged from 0.4 to 4.8 dB. The performance of bilateral subjects on average is closer to that of normal hearing individuals, but is still at least 3 dB poorer. Differences in noise level, type, and location between studies could account for some of the variance, but it is not likely to account for all of the variation. ASC and BEAM processing improve the ability of CI users to understand speech in background noise, but performance with these strategies is far from that of normal hearing individuals.

Future research should focus on increasing the sample size of bilateral subjects. The small sample of three subjects on average revealed a bilateral improvement, but no statistical analyses could be completed to show the significance of this improvement. Bilateral implantation is becoming more common, which may help bilateral subject recruitment in a future study. Also, combinations of two or three processing options are available in the current Cochlear programming software. Possible combinations include: ADRO and ASC, ADRO and BEAM, ASC and BEAM, and ADRO, ASC, and BEAM. Future studies should explore performance with multiple processing options active. In combination, these strategies could complement each other to further improve speech recognition in noise. However, added processing may also distort the speech signal. Removing noise from the input signal could lead to the elimination of speech information necessary for accurate understanding. Finally, many CI recipients also use a hearing aid on the contralateral ear (bimodal listening), and the hearing aids often employ directional microphones to improve speech recognition in noise. Further research should investigate the performance of these CI processing options with bimodal users. Bimodal listening might provide sufficient bilateral input to improve speech recognition in noise. The bimodal benefit may be dependent on the amount of residual hearing of the CI recipient, as well as the type of hearing aid (i.e. if the hearing aid has a directional microphone, type of noise processing, etc.). The hearing aid may complement the CI to improve speech recognition in noise, or the hearing aid could simply amplify the noise negatively affecting speech recognition.

CONCLUSIONS

The results obtained from unilateral CI subjects suggest the use of processing options that utilize noise reduction processing, like that available in ASC and BEAM, improves a CI recipient's ability to understand speech in noise in listening situations similar to those experienced in the real-world. The choice of the best processing option is dependent on the noise level, with BEAM best at moderate noise levels and ASC best at loud noise levels. The ability of these speech processing options to improve speech recognition in noise has been documented in previous research, but the R-Space system has not been used in earlier work and may better predict the true benefit provided by these processing options. Although ASC and BEAM processing have demonstrated the ability to improve speech recognition in noise for CI recipients, their performance is still poorer than that observed in normal hearing individuals.

Results for three bilateral CI subjects show a bilateral improvement in speech recognition in noise when compared to the better ear alone. This benefit can most likely be attributed to the effects of binaural squelch and redundancy due to the diffuse noise source of the R-Space. The bilateral improvement increases as the noise level increases, suggesting a more significant bilateral benefit in more challenging listening situations. These findings support the use of processing options that utilize noise reduction to improve speech recognition in noise for unilateral and bilateral CI recipients. The addition of these options should be part of the standard programming protocol to increase CI recipient satisfaction and benefit. Lastly, counseling the CI recipient on the adjustment of the processor depending on the environment and noise level should be done routinely.

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Figure 1: BEAM preprocessing scheme. Figure taken from Wouters and Vanden Berghe (2001) and used with permission from Jan Wouters.

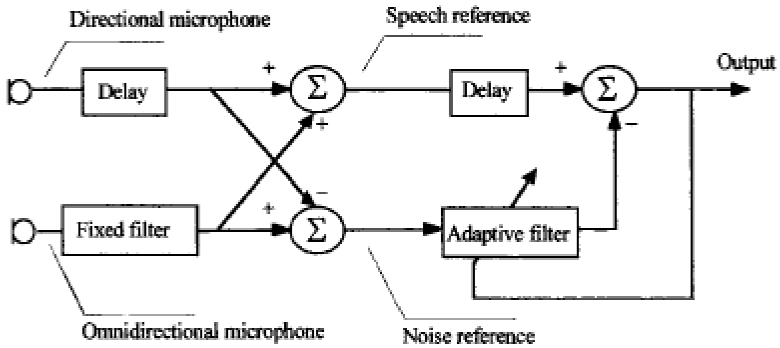

Figure 2: Directional polar plots comparing standard hardware directional microphone and BEAM. Signal is at 0° azimuth and noise source was located at 90°, 180°, and 270° azimuths. Concentric circles are in units of 5 dB. Figure taken from Patrick et al (2006) and used with permission from James Patrick.

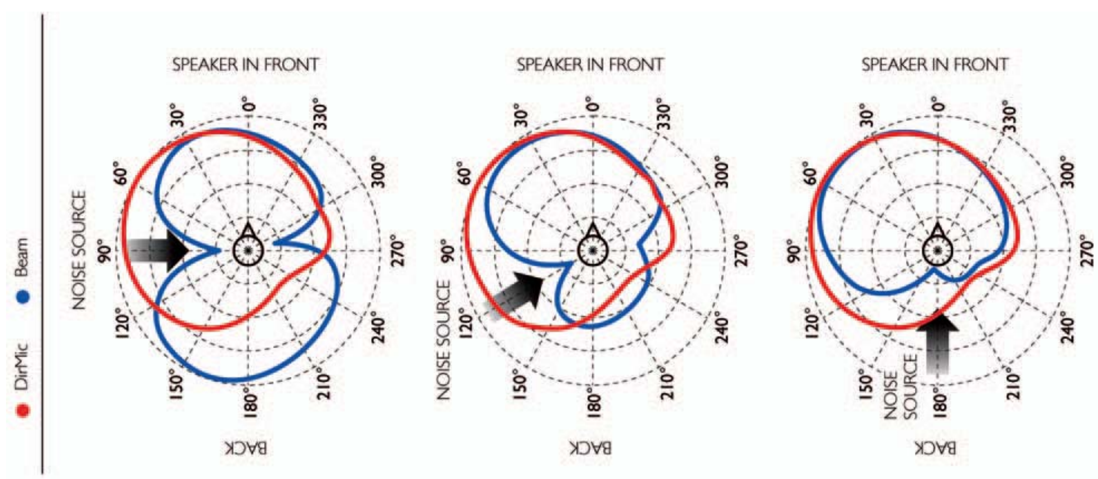

Table 1: Unilateral subjects' demographic and hearing history information. Bilateral subjects are denoted in bold.

Subject	Gender	Age	Implanted Ear	Years of HL	Years of Severe to Profound HL	Years of HA Use	Etiology
1	M	32	L	31	31	28	Unknown
2	**F**	**50**	**R**	**40**	**20**	**29**	**Genetic**
3	F	45	R	18	14	14	Unknown
4	M	37	R	36	36	33	Maternal Rubella
5	M	58	L	9	2	7	Unknown
6	M	48	L	39	35	39	Genetic
7	M	67	L	35	3	10	Noise exposure
8	**M**	**65**	**R**	**54**	**2**	**24**	**Measles**
9	**M**	**50**	**L**	**47**	**43**	**47**	**Unknown**
10	F	75	R	30	4	30	Unknown
11	F	40	R	32	3	0	Unknown
12	F	80	L	20	15	10	Unknown
13	M	75	L	44	14	38	Otosclerosis
14	F	68	R	30	5	20	Measles
15	F	46	R	37	35	37	Measles
16	F	77	R	11	1	7	Unknown
17	M	82	L	40	25	20	Ototoxicity
18	M	71	L	40	5	35	Otosclerosis
19	F	25	L	6	3	3	Meniere's Disease
20	M	50	R	48	8	37	Maternal Rubella
21	M	68	L	49	15	30	Noise exposure
22	F	78	R	1	1	1	Unknown
23	M	58	R	21	3	19	Unknown
24	F	70	L	15	6	0	Meniere's Disease
25	F	60	L	45	10	15	Otosclerosis
26	M	49	R	45	45	39	Meningitis
27	M	78	R	25	4	4	Unknown
28	F	57	L	30	5	14	Genetic
29	M	70	L	22	20	1	Ototoxicity
30	M	61	L	22	1	17	Unknown
Mean		60		30.7	13.8	20.3	
SD		15.2		13.8	13.9	14.2	

Table 2: Unilateral subjects' programming information. Bilateral subjects are denoted in bold.

Subject	Implanted Ear	Volume	Sensitivity	Strategy	Rate (Hz)	Maxima	Years of Implant Use	CNC Score
1	L	7	12	ACE	2400	6	1	19%
2	**R**	**7**	**12**	**ACE**	**1800**	**8**	**5**	**58%**
3	R	7	12	ACE	900	8	3	57%
4	R	7	11	ACE	500	10	1	25%
5	L	9	14	ACE	1200	10	3	69%
6	L	9	12	ACE	500	10	3	36%
7	L	9	12	ACE	1800	8	5	74%
8	**R**	**8**	**11**	**ACE**	**900**	**12**	**6**	**82%**
9	**L**	**9**	**12**	**ACE**	**1800**	**8**	**3**	**55%**
10	R	9	12	ACE	900	8	6 months	41%
11	R	7	10	ACE	1800	10	3	86%
12	L	7	12	ACE	1200	10	5	63%
13	L	7	14	ACE	1800	8	5	80%
14	R	7	12	ACE	1800	10	3	52%
15	R	7	12	ACE	1800	8	4	46%
16	R	7	12	ACE	1200	10	5	72%
17	L	7	11	ACE	2400	10	2	17%
18	L	9	9	ACE	1800	8	4	50%
19	L	7	10	ACE	1800	8	6 months	52%
20	R	9	12	ACE	2400	10	3	75%
21	L	7	12	CIS	900	10	6 months	24%
22	R	6	8	ACE	1800	8	8	48%
23	R	7	12	SPEAK	250	8	4	82%
24	L	7	12	ACE	1200	8	2	57%
25	L	9	11	ACE	900	12	6	78%
26	R	7	11	ACE	1200	8	3	60%
27	R	7	12	ACE	2400	10	2	46%
28	L	7	12	MP 3000	500	6	1	52%
29	L	7	10	ACE	1200	10	6 months	58%
30	R	7	11	ACE	1800	8	1	85%
Mean							3.4	56.8%
SD							2.0	20%

Table 3: Hearing history and programming information of bilateral subjects for the ear not included in the unilateral analysis.

Subject	Implanted Ear	Years of HL	Years of Severe-Profound HL	Years of Hearing Aid Use	Years of Implant Use	Years Between Implants	Years of Bilateral Use
2	L	40	20	32	3.4	2	3.4
8	L	54	2	5	1.7	4.5	1.7
9	R	43	39	43	6.5	3.5	3
Mean		45.7	20.3	26.7	3.9	3.3	2.7
SD		6.0	15.1	16.0	2.0	1.0	0.8

Subject	Volume	Sensitivity	Strategy	Rate (Hz)	Maxima	CNC Score
2	7	12	ACE	1800	8	62%
8	8	11	ACE	1800	8	85%
9	9	12	ACE	1800	8	52%
Mean						66.3%
SD						13.8%

Figure 3: R-Space Array. Figure taken from Compton-Conley et al (2004) and used with permission from Cynthia Conley.

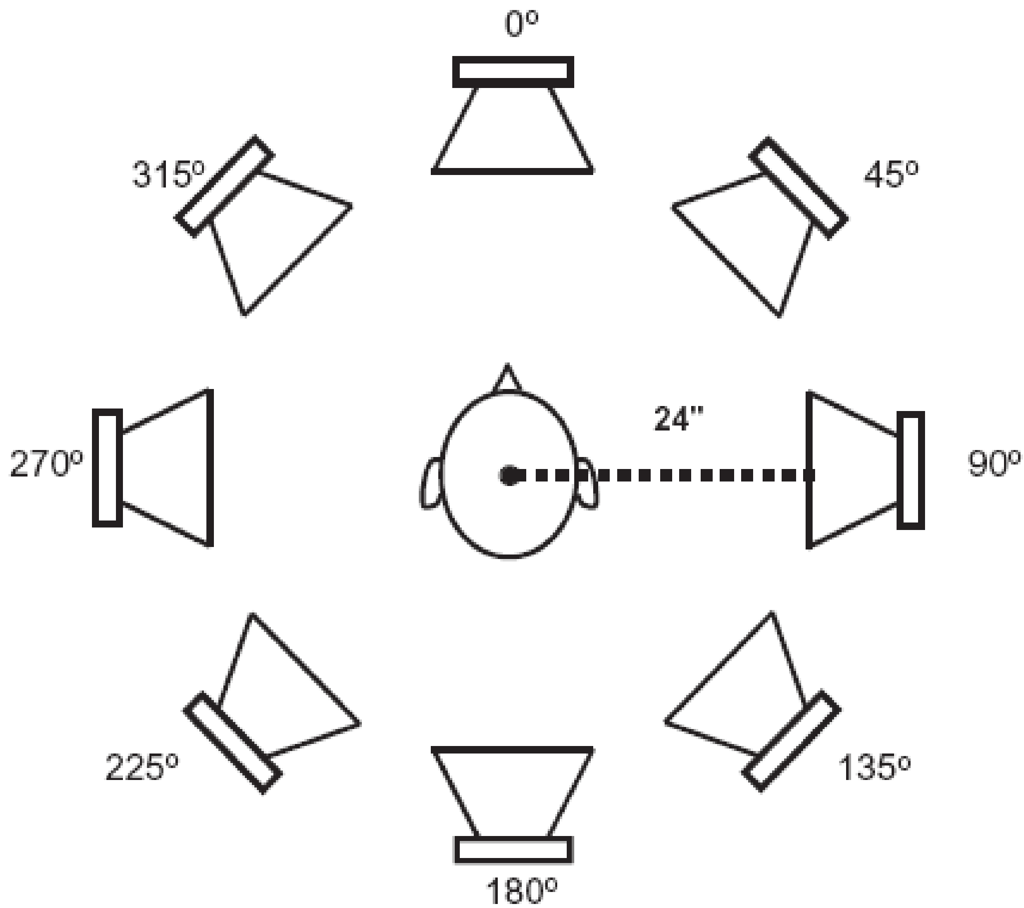

Figure 4: Mean soundfield thresholds and +/- 1 standard deviation with CI of unilateral subjects

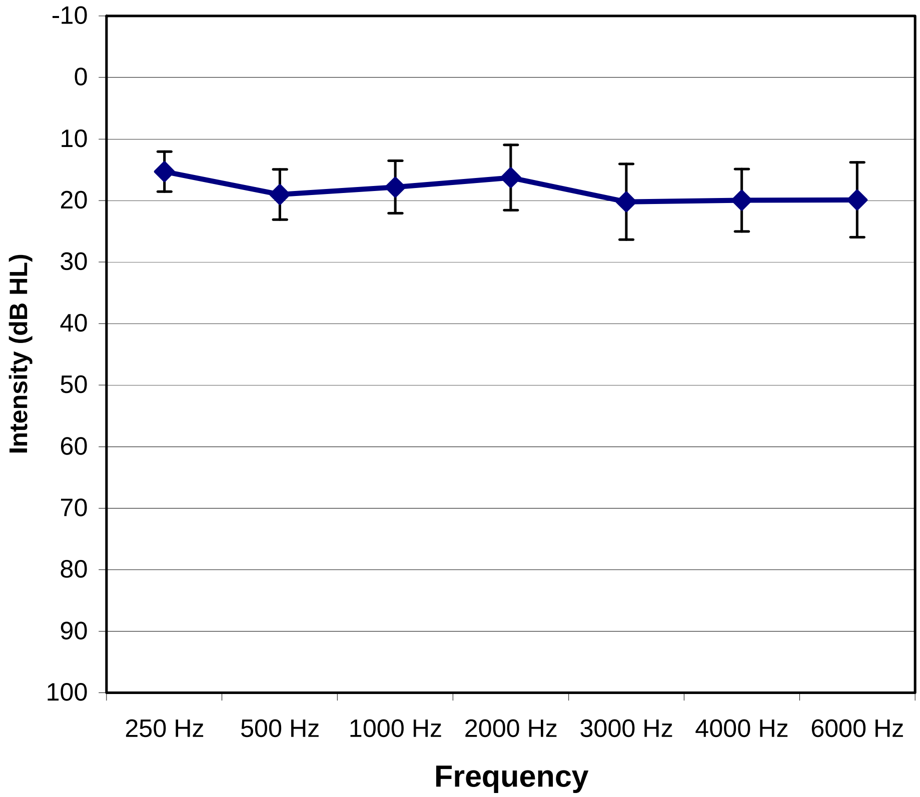

Figure 5: Mean RTS of unilateral subjects in 60 dB SPL noise for STD, ADRO, ASC, and BEAM processing conditions. Error bars represent +1 standard deviation and statistical significance ($p \leq 0.05$) is indicated by the *.

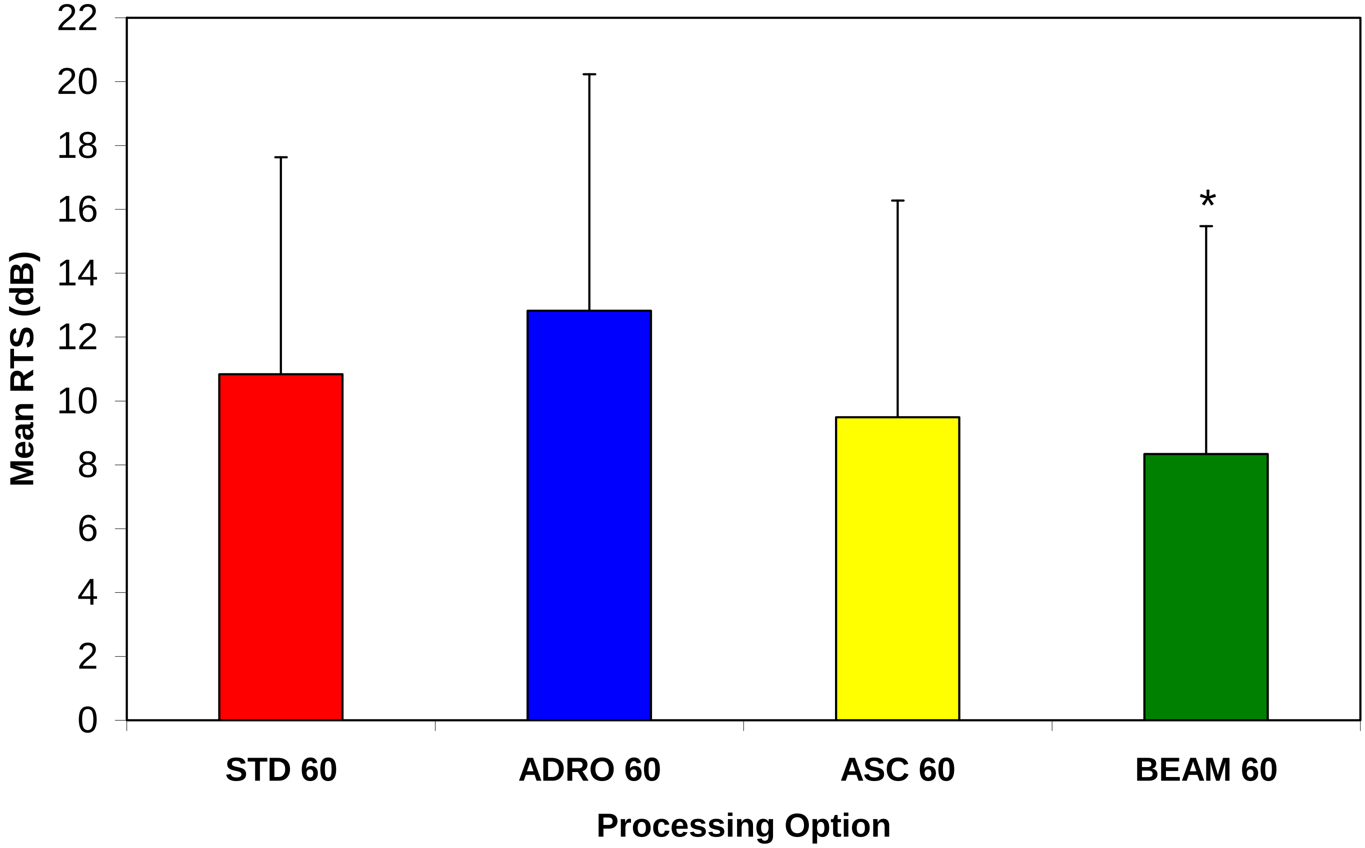

Figure 6: Mean RTS of unilateral subjects in 70 dB SPL noise for STD, ADRO, ASC, and BEAM processing. Error bars represent +1 standard deviation and statistical significance ($p \leq 0.05$) is indicated by the *.

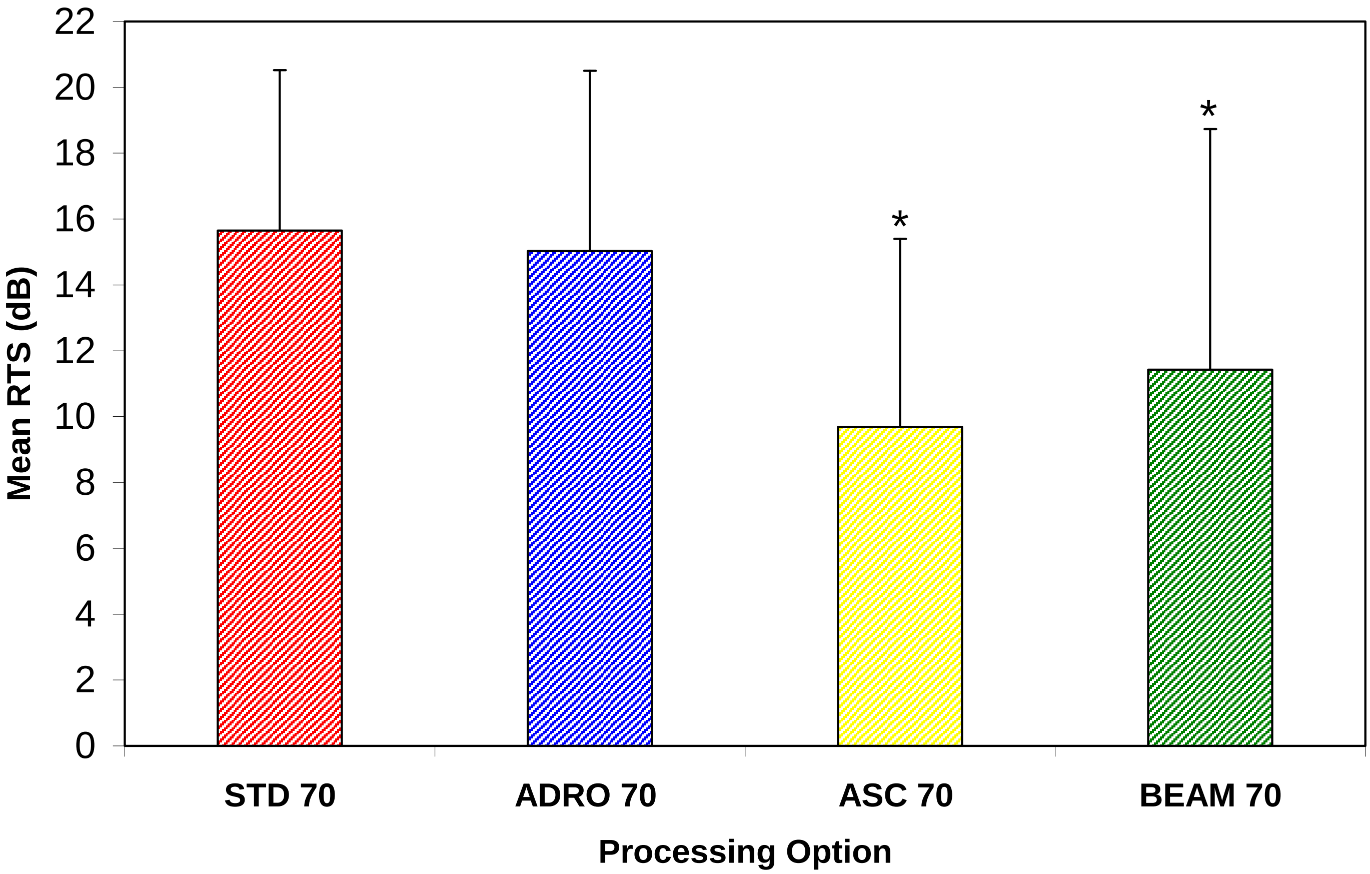

Figure 7: Mean RTS difference between 60 and 70 dB SPL noise of unilateral subjects (RTS at 70 dB SPL – RTS at 60 dB SPL) for STD, ADRO, ASC, and BEAM processing. Error bars represent +1 standard deviation and statistical significance (p≤0.05) is indicated by the *.

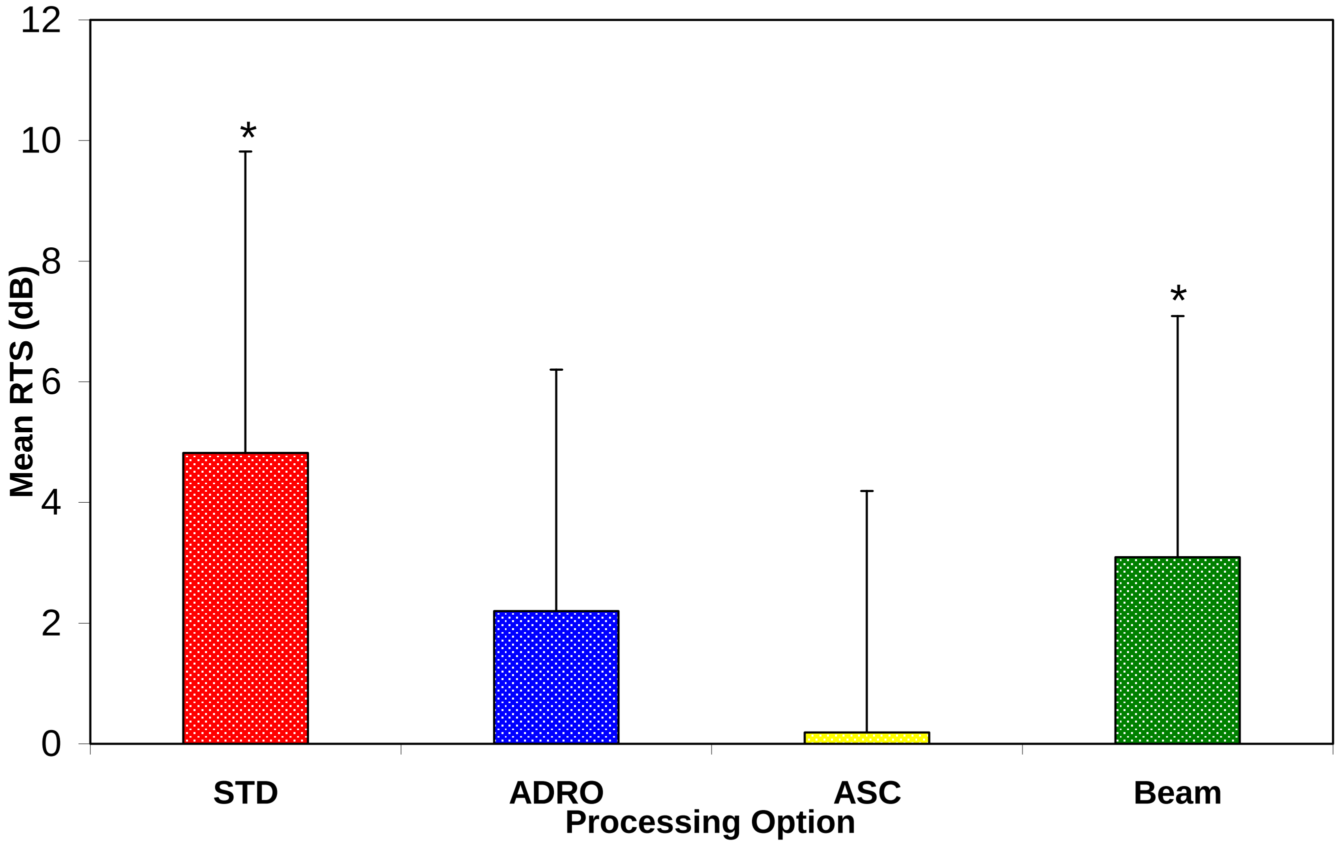

Figure 8: Mean RTS of bilateral subjects in 60 dB SPL noise for STD, ADRO, ASC, and BEAM processing. Mean RTSs are shown for right and left unilateral and bilateral conditions.

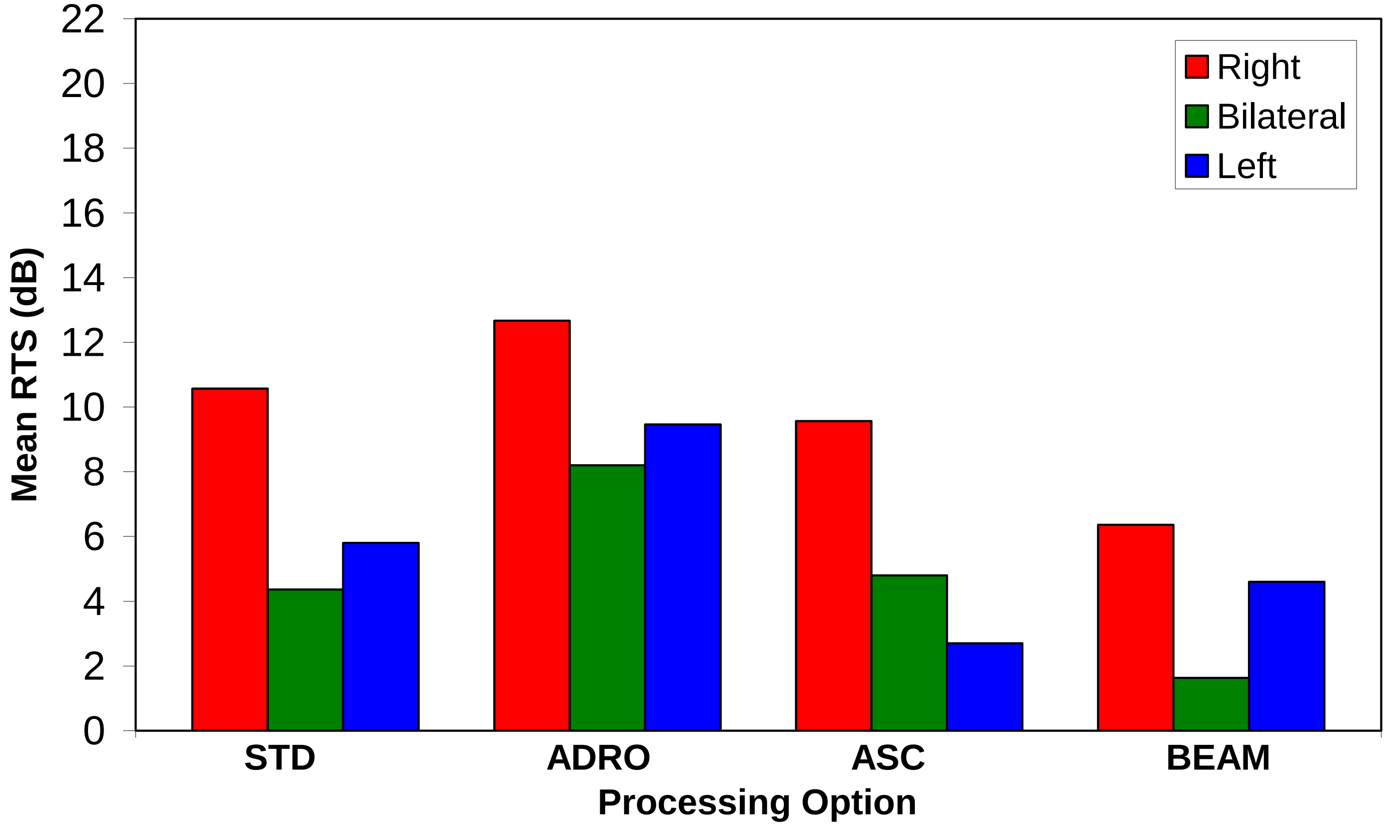

Figure 9: Mean RTS of bilateral subjects in 70 dB SPL noise for STD, ADRO, ASC, and BEAM processing. Mean RTSs for right and left unilateral and bilateral conditions are shown.