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Evaluation of multiple speech processing combinations in the Cochlear Nucleus 5 cochlear implant system using R-Space simulation

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EVALUATION OF MULTIPLE SPEECH PROCESSING COMBINATIONS IN THE COCHLEAR NUCLEUS 5 COCHLEAR IMPLANT SYSTEM USING R-SPACE SIMULATION

by

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Abstract: Difficulty understanding speech in the presence of background noise is a common report among cochlear implant recipients. The purpose of this research is to evaluate speech processing options currently available in the Cochlear Nucleus 5 sound processor to determine the best option for improving speech recognition in noise.

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INTRODUCTION AND REVIEW OF THE LITERATURE

Advances in cochlear implant technology have allowed most cochlear implant recipients to achieve substantial levels of speech recognition (Dowell et al., 1986; McKay & McDermott, 1993; Fishman et al., 1997; Skinner et al., 1997; Balkany et al., 2007). For example, Balkany and colleagues (2007) measured speech recognition performance in a group of adult Nucleus Freedom cochlear implant (CI) recipients. Preoperative speech recognition was assessed using the Hearing in Noise Test (HINT) sentences presented in quiet at 70 dB SPL and consonant-nucleus-consonant (CNC) words presented in quiet at 60 dB SPL. Mean preoperative HINT sentence and CNC word scores were 11.3% and 3.0% respectively, although a wide distribution of preoperative scores was observed. Over 40% of the participants had no open-set sentence recognition preoperatively, and only four scored over 30% on HINT sentences in quiet. Postoperative speech recognition that was tested after six months of CI use revealed significant improvements in speech recognition. Mean postoperative scores for HINT sentences and CNC words in quiet were 78% and 57% respectively. Although advances in cochlear implant technology offer opportunities for improved speech recognition, CI recipients may still encounter much greater difficulty when listening in background noise.

Nelson and colleagues (2002) reported CI recipients are more susceptible to the effects of background noise compared to normal-hearing listeners due to their inability to take advantage of temporal gaps in fluctuating noise. Release from masking was examined for CI recipients, normal-hearing listeners responding to implant simulations, and normal-hearing controls. All participants repeated sentences in quiet, steady-state noise, and modulated speech-weighted noise. The modulated noise was presented at varying SNRs at various modulation rates. Release from masking was measured as percent improvement in scores for modulated versus steady noise. Masking release results were compared at different SNRs: -8 dB SNR for the normal-hearing group and +8 dB SNR for the CI and simulation groups. Results revealed that both the CI and simulation groups experienced very little release from masking from modulated maskers compared to the normal-hearing listeners. At a SNR of -8 dB, the normal-hearing group obtained significant release from masking, with a 60-80% mean improvement in performance from the modulated (4 Hz or higher) versus steady noise. Unlike the normal-hearing group, the CI and simulation groups obtained very little benefit from the modulated maskers. At a +8 dB SNR, the simulation group showed no masking release at any modulation rate. For the CI recipients, a slight release from masking was observed at extremely slow (1 Hz) and fast (16 and 32 Hz) modulation rates. The inability of the CI and simulation groups to take advantage of temporal gaps in background noise was thought to be attributed to their limited spectral resolution and/or the use of envelope-based signal processing strategies.

Notable differences in speech recognition performance are reported in the literature for CI recipients tested in noisy versus quiet conditions. Firszt and colleagues (2004) reported that CI recipients' performance on sentence recognition tasks was significantly poorer with the introduction of noise compared with listening at a soft conversational level in quiet. An average decrease of 16% was observed between HINT sentences presented at 60 and 50 dB SPL in quiet, while a mean decrease of 30% was found between sentences presented at 60 dB SPL in quiet versus 60 dB SPL in the presence of noise at a +8 dB SNR. Fetterman and Domico (2002) also found significant decreases in CI recipients' sentence recognition scores with the addition of background noise. On average, participants scored 82% correct on City University of New York (CUNY) sentences in quiet. With increasing levels of background noise, mean performance decreased to 73% (at a +10 dB SNR) and 47% (at a +5 dB SNR).

Improvements in technology now provide signal processing options that may improve speech recognition, specifically in noise. The terms preprocessing, input signal processing, or front-end processing are used interchangeably in reference to the signal processing that is carried out in the external speech processor. The processing takes place prior to the band-pass filter stage, so it is considered part of the preprocessing stage before the signal is sent to the filter bank. Several speech processing options have been introduced, over the years, by Cochlear Americas, to further improve speech recognition in quiet and in noise, including Adaptive Dynamic Range Optimization (ADRO), Autosensitivity (ASC), and Beam. An additional processing option known as Zoom became available in the newest speech processor, the CP810.

These processing options function to modify the amount of stimulation received by the CI recipient to optimize speech recognition performance. They are designed to aid the process in which a wide range of acoustic inputs is programmed into the recipient's narrow range of electrical stimulation. Cochlear uses an instantaneous input dynamic range (IIDR) to map an acoustic input signal onto an electrical output signal. The IIDR determines the range of the acoustic input signal that is processed at any given point in time and programmed without compression into the electrical dynamic range (EDR). The EDR is the range of electrical stimulation that is perceived by the listener at each electrode (Wolfe, 2010). Together, threshold (T) and maximum comfort (C) levels determine the dynamic range of electrical stimulation for each electrode channel.

In the Advanced Combination Encoder (ACE) and Spectral Peak (SPEAK) speech coding strategies, a channel gain is applied to the output of the filter for each channel to produce a channel amplitude. The maximum channel amplitude corresponds to a 0 dB reference level, and sounds mapped at this level produce C-level electrical stimulation. Since the default IIDR for the Nucleus Freedom and Nucleus 5 sound processors is 40 dB SPL, channel amplitudes corresponding to -40 dB produce stimulation at T-level. Amplitudes below -40 dB do not produce any stimulation (James et al., 2002). In ACE and SPEAK, stimulation is only applied to electrodes that correspond to the channels with the largest amplitudes, or maxima. For a program with no additional input processing, the channel gain is fixed. However, for programs utilizing Adaptive Dynamic Range Optimization (ADRO) processing, the channel gains are continually adjusted to match the input signal to specific targets in the upper part of the 40 dB IIDR. Several statistical rules are applied independently in each channel to keep the output level between a comfort and audibility target. These rules use percentile estimates of the channel amplitudes (James et al., 2002). Sounds below the audibility target are increased while sounds above the comfort target are decreased. By adjusting the input signal so that the output is comfortably loud, ADRO aims to improve speech recognition at low and medium levels and reduce loudness discomfort for loud sounds (James et al., 2002).

Several studies have investigated the benefit of ADRO processing in cochlear implants. Dawson and colleagues (2004) evaluated speech recognition in quiet and in noise in 15 children using the SPrint processor. Sentence recognition was assessed with the child's standard everyday program with no additional processing and the ADRO program using BKB sentences presented at 50 dB SPL in quiet and at 65 dB SPL in the presence of 8-talker babble noise. To avoid ceiling effects, the SNRs were selected on an individual basis and ranged between 0 and +15 dB. ADRO showed significant improvement over the standard program in both quiet and in noise. In quiet, mean performance with ADRO was 8.6% higher compared with the standard program. In noise, ADRO showed a mean improvement of 6.9% over the standard program. James and colleagues (2002) evaluated speech perception with and without ADRO processing in

9 adult cochlear implant recipients using the SPrint processor. CUNY sentences were presented in quiet and in 8-talker babble noise at +10 and +15 dB SNR. For sentences presented in quiet at 50 dB SPL, performance with ADRO revealed a significant mean improvement of 16% compared to the standard program. No significant difference in sentence scores was found among the ADRO and standard programs in either of the noise conditions. While both of these studies demonstrate the benefits of ADRO in quiet, the ability of ADRO to improve speech recognition in noise yielded mixed results.

Microphone sensitivity influences the position of the IIDR by determining the input signal level needed to produce C- and T-level stimulation (Cochlear Limited, 2007). Acoustic input levels within this range are mapped without compression into the patient's EDR. Changes in microphone sensitivity alter the softest level sound that is mapped the EDR. This also alters the automatic gain control (AGC) kneepoint, or the input level corresponding to C-level stimulation, above which sounds are infinitely compressed. An increase in sensitivity lowers the input level that will be converted into T-level stimulation mapping softer sounds into the electrical dynamic range. This also reduces the AGC kneepoint, the input level subjected to compression, so louder sounds will be compressed more. In contrast, decreasing sensitivity increases the input level at which the recipient receives T-level stimulation. This may affect audibility for soft sounds, but it may improve speech recognition in noise by reducing amplification of low-level background noise (Wolfe, 2009). Another input processing strategy available in Cochlear speech processors is Autosensitivity (ASC). ASC is designed to automatically adjust the microphone sensitivity depending on the noise level and the signal-to-noise ratio at the speech processor microphone. ASC monitors the noise floor by analyzing troughs in the envelope of the input signal. The level of the speech signal is estimated when the modulation rate of the incoming signal is characteristic to that of typical speech. During breaks in speech, an estimate of the noise level is obtained. In quiet, a program with ASC acts similar to a program with no additional input processing, with a fixed sensitivity setting of 12. If the level of background noise is above 57 dB, sensitivity is reduced according to the level of the noise so that the peaks of speech exceed the long-term average noise spectrum by at least 15 dB (Wolfe, 2009).

For listening to high level speech in moderate to high-level noise, reducing sensitivity will prevent compression of the speech and the user's EDR will contain a larger portion of the desired signal. In addition, less ambient noise will be mapped to the EDR. Thus, ASC is a recommended strategy for improving speech recognition in environments with high levels of noise (Wolfe, 2010).

In 2005, a two-microphone adaptive directional processing strategy referred to as Beam was incorporated into Cochlear's Nucleus Freedom speech processor (Wouters and Vanden Berghe, 2001). Beam is a single spectral channel adaptive filtering directional processing scheme designed to produce an adaptive directional response with maximum sensitivity at 0º and maximum suppression between 90º and 270º azimuth (Cochlear Limited, 2010a). Beam utilizes a directional microphone and an omnidirectional microphone in a two-stage process designed to improve the signal-to-noise ratio by focusing on sounds arriving from the front. The first stage is a spatial preprocessing stage, in which Beam utilizes two directional patterns, one facing forward and one facing backwards, to create a speech reference and a noise reference. Beam searches for the loudest noise source to create the noise reference. This is then subtracted from the forward facing speech reference to achieve the desired directional filtering (Cochlear Support, 2011). The second stage uses an adaptive noise cancellation process to reduce the residual noise in the speech reference. To limit distortion of speech, the adaptive noise cancellation stage adapts only during breaks in speech (Wouters & Vanden Berghe, 2001).

Spriet and colleagues (2007) evaluated speech performance in noise with Beam processing compared to no additional processing with the standard directional microphone in five adult cochlear implant recipients using the Nucleus Freedom processor. Two different noise source configurations were used: a single noise source at 90º, and three noise sources at 90º, 180º, and 270º. Adaptive SRT measurements were made using both speech-weighted noise and multitalker babble noise in each of the noise configurations. Beam demonstrated significant improvement over the standard directional microphone in all noise conditions. For a single noise source at 65 dB SPL, a mean SNR improvement of 13.4 dB and 15.9 dB was found for speech-weighted and multitalker babble noise, respectively. For the three noise source condition, the mean SNR improvement for speech weighted noise was 6.5 dB; for multitalker babble noise, the average SNR improvement was 11.6 dB. The results of this study demonstrate the benefit of Beam processing versus the standard directional microphone for speech perception in noise.

While test measures in the booth can help demonstrate the efficacy of input processing strategies, they may not provide an accurate representation of the benefit obtained in everyday listening environments. Several studies suggest that it may be difficult to predict real-world performance with directional microphones due to environmental variations such as reverberation, SNR, intensity of the speech signal, location of the listener, and location of the noise source(s) (Hawkins & Yacullo, 1984; Surr et al., 2002; Walden et al., 2003; Dittberner & Bentler, 2007; Gnewikow et al., 2009). All of these factors are capable of confounding any directional benefit measured in the test booth.

In an attempt to more accurately reproduce real-world listening environments in a sound booth, Compton-Conley and colleagues (2004) created the R-Space test system, a simulation utilizing recorded restaurant noise played through eight loudspeakers arranged in a circular pattern around the listener. To assess whether or not real-world performance of directional microphone hearing aids could be accurately tested in the test booth, measurements were performed using directional microphones in three noise conditions: the R-Space, restaurant noise from a single speaker 90º directly above the listener, and restaurant noise from a single speaker 180º behind the listener. Each of the three simulation techniques was then compared to measurements recorded at a live restaurant (live condition). The R-Space was the only condition in which performance was not significantly different from the live condition (Compton-Conley et al., 2004). Since the R-Space is more effective in simulating real-world background noise compared to other noise configurations commonly implemented in the sound booth, measurements made using the R-Space may provide a more accurate assessment of directional benefit. Results from this study support the use of the R-Space to evaluate the benefit of processing options used in cochlear implants.

Brockmeyer and Potts (2011) measured speech recognition of adult Freedom recipients in the R-Space with four processing options: a standard dual-port directional microphone, ADRO, ASC, and Beam. Participants repeated HINT sentences presented at 0º azimuth with R-Space noise at 60 and 70 dB SPL. An adaptive reception threshold for sentences (RTS) was measured for each processing condition at both noise levels. At 60 dB SPL noise level, Beam processing provided the best mean RTS. At 70 dB SPL noise, ASC demonstrated the best mean RTS, although both ASC and Beam demonstrated significantly better mean RTS scores compared to the standard or ADRO programs. Results from this study suggest that real-world benefit from specific processing options varies as a function of noise level. Thus multiple processing options may be required to provide maximum benefit to cochlear implant recipients in a variety of listening environments (Brockmeyer & Potts, 2011).

Gifford and Revit (2010) evaluated speech recognition performance of 20 adult Freedom recipients utilizing their everyday preferred program with no additional processing compared to a program with a combination of processing options, specifically Beam+ASC+ADRO. All participants were Freedom recipients and their preferred programs had either ASC, ADRO, or ASC+ADRO active. HINT sentence recognition was measured using an adaptive procedure in the R-Space with a fixed noise level of 72 dB SPL. Mean SRT performance for the everyday preferred program and Beam+ASC+ADRO was 11.2 and 7.3 dB SNR, respectively. Beam+ASC+ADRO resulted in equal or better performance for all participants, with a mean degree of improvement in SRT of 3.9 dB. In response to a poll in which 18 of the 20 participants reported they did not switch programs in noise, a second experiment was designed to determine whether ASC+ADRO might be a better option for an everyday program compared to ADRO alone. ASC was omitted with the rationale that participants using ASC alone as their everyday program were less likely to notice a difference with the addition of Beam as compared to those using ADRO alone. SRTs using ADRO, ASC+ADRO, and Beam+ASC+ADRO were obtained under similar conditions used in the previous experiment. The ASC+ADRO condition resulted in a mean improvement of 2.5 dB as compared to ADRO alone. Beam+ASC+ADRO resulted in the lowest, or best, SRT with mean improvements from ADRO alone and ASC+ADRO of 6.1 dB and 3.6 dB, respectively. Thus, the authors suggest using ASC+ADRO as the default everyday program and Beam+ASC+ADRO for noisy environments.

Further advancements in technology have led to the release of the Cochlear Nucleus 5 Sound Processor (CP810) by Cochlear Americas in September 2009. Unlike the Freedom sound processor that uses an omnidirectional microphone plus a hardware dual-port directional microphone, the CP810 uses two omnidirectional microphones which can be combined to produce several directional responses. In every CP810 sound processor, the microphones are calibrated by Cochlear using +/- 1 dB tolerance to confirm identical gain and phase responses (Cochlear Support, 2011). The choices for directionality available in the CP810 processor include omnidirectional, Beam, and Zoom (Cochlear Limited, 2010a).

The Beam algorithm in the CP810 sound processor differs from the Beam setting in the Freedom because it utilizes two omnidirectional microphones to create the speech and noise references. The speech reference in the Freedom is the front directional microphone output, which produces a slightly forward facing directional pattern. A combination of the outputs from the front and rear microphones is utilized to form a rear facing directional pattern for the noise reference. In the CP810, the two omnidirectional microphones are utilized to create two strongly directional patterns (forward and rear facing) for the speech and noise references. Similar to the Freedom, the CP810 Beam utilizes an additional adaptive noise cancellation stage to reduce the remaining noise in the speech reference. The main difference in Beam between the two processors is that the CP810 dual omnidirectional microphones are easier to control and can more accurately tune the directional patterns to create an optimal speech and noise reference (Cochlear Support, 2011). The CP810 Beam is expected to have about a 5 dB increase in attenuation compared to the Freedom Beam (Cochlear Limited, 2010a).

To address situations in which a more fixed pattern of directionality is desirable, Cochlear introduced Zoom into CP810 sound processor. Zoom produces a fixed hypercardiod pattern through a combination of the dual omnidirectional microphone signals. Zoom is designed to reduce sounds behind and to the sides of the listener with maximum suppression at ±120º azimuth (Cochlear Limited, 2010a). Similar to Beam, Zoom utilizes a spatial preprocessing stage to create a speech and a noise reference for directional filtering. However, Zoom does not perform a second adaptive noise cancellation to reduce the remaining noise in the speech reference.

Recent data were gathered from an Australian clinical validation study sponsored by Cochlear evaluating speech perception in noise with the Freedom and CP810 sound processors. A group of 19 adult Freedom cochlear implant recipients were evaluated using Beam processing with the Freedom and the CP810 sound processors. Performance was assessed as percentage correct on CUNY sentences presented at 65 dB SPL in noise. Four-talker babble was delivered simultaneously from loudspeakers at 90º, 180º and 270º, and SNR was optimized to keep scores between 30-70% to avoid floor and ceiling effects. Participants were tested using their preferred Beam setting. Group mean scores were 67% with Beam enabled in the CP810 and 55% with Beam enabled in the Freedom. This 12% improvement in group mean performance suggests improved performance with the Beam in the CP810 processor (Cochlear Limited, 2010b).

A second study incorporated the same test procedures but compared performance using the standard directional mode with both the Freedom versus CP810 sound processors, and also with Zoom enabled on the CP810 sound processor. No significant difference was found between performance using the standard directional mode with the Freedom and CP810 processors, without Zoom enabled on the CP810. However, group mean results revealed an average 74% correct with Zoom enabled compared to 44% with Zoom disabled in the CP810 sound processor (Cochlear Limited, 2010b).

The default setting for everyday listening in Cochlear's most recent software, Custom Sound Suite 3.2 is ASC+ADRO. Beam+ASC+ADRO is recommended for situations in which the noise source is moving or if there are discrete noise sources that are louder at different points in time. Zoom+ASC+ADRO is recommended when the desired signal is relatively stationary and in front of the listener and the noise is behind the listener and not moving. For quiet and moderately noisy environments when there is no specific sound source, the standard fixed directional setting is recommended (Cochlear Limited, 2010a).

To date, the only data evaluating the benefits of the front-end processing strategies in the CP810 sound processor is preliminary data from validation trials sponsored by Cochlear Americas. Performance with these processing options has not yet been evaluated in an environment resembling real-world listening conditions. The purpose of the current study was to evaluate the processing options currently available in the CP810 sound processor using the R-Space speaker setup to determine which processing option(s) performs best in R-Space background noise. Eight processing options were evaluated including Beam-only, Beam+ASC, Beam+ADRO, Beam+ASC+ADRO, Zoom-only, Zoom+ASC, Zoom+ADRO, and Zoom+ASC+ADRO. This study could help determine which processing option results in better speech recognition in background noise. The results of this study may have implications for programming decisions that result in increased benefit for cochlear implant recipients listening in background noise.

METHODS

Participants

Thirty-two adult cochlear implant recipients participated in the study. Participants included 14 females and 18 males, ranging in age from 36 to 92 years, with a mean age of 66 years. Table 1 contains individual and mean demographic and audiologic information. Duration of hearing loss prior to implantation ranged from 9 to 57 years, with a mean duration of 33 years. The mean duration of severe-to-profound hearing loss prior to implantation was 10 years, with a range of 1 to 45 years. Years of hearing aid use prior to implantation ranged from 0 to 48, with a mean duration of 20 years.

Participants were implanted with the Cochlear Contour Advance (CI24RE) or Nucleus 5 (CI512) internal cochlear implant arrays. Table 2 contains information related to cochlear implant use. Of the 32 participants, 25 used Cochlear Nucleus Freedom sound processors and 7 used Nucleus 5 (CP810) sound processors. The mean duration of cochlear implant use was 3.1 years, with a range of 0.7 to 6.9 years. All participants used the ACE speech coding strategy. Use of input processing varied among the participants' everyday preferred programs. Thirteen preferred programs used ADRO, 10 did not use input processing, 3 used ASC+ADRO, 2 used ASC only, 2 used Beam+ASC+ADRO, 1 used Beam+ASC, and 1 used Beam-only.

All participants were recruited from the patient population of the Washington University School of Medicine Department of Otolaryngology and programmed following a specified clinical protocol (Skinner et al., 1995; Sun et al., 1998; Skinner et al., 1999, Holden et al., 2002; Skinner et al., 2002). To ensure that participants had measurable open-set speech recognition, only those who had scores greater than 20% on CNC word recognition at their most recent clinical evaluation were recruited. Six of the participants were bilateral CI recipients and the test ear was chosen at random if both ears met the criteria for inclusion.

Approval for this study (#10-1164) was obtained from the Washington University School of Medicine Human Research Protection Office (HRPO) prior to data collection. Participants signed an informed consent document approved by the HRPO committee. Participants were reimbursed for their time and travel.

Equipment/Test Environment

The Cochlear Nucleus Freedom Contour Advance (CI24RE) and Nucleus 5 (CI512) internal cochlear implants consist of a receiver/stimulator, a 22-electrode array, and 2 extracochlear electrodes. Both internal devices have the same electrode array, but the receiver stimulator in the System 5 is slightly thinner (3.9 mm vs. 6.9 mm) and lighter (8.8 g vs. 9.5 g) (Cochlear Limited, 2005; Cochlear Limited, 2009). Together the CI512 internal implant and the CP810 external sound processor constitute the Nucleus 5 System. The CP810 sound processor is currently compatible with both the Freedom and Nucleus 5 cochlear implants.

Four of the eight processing options tested in this study utilize Zoom and are only available in the CP810 sound processor. All participants were tested using a loaner CP810 processor programmed using Custom Sound v3.0 developed by Cochlear Americas. The processor was hardwired to a programming interface (programming POD) connected to a personal Hewlett-Packard computer equipped with the programming software. For the twenty-five participants utilizing a Freedom processor in everyday life, a CP810 processor program was created for testing with the CP810 processor. The individual's Freedom processor program was converted to an equivalent CP810 program, and the appropriate processing options were added. If the preferred program utilized a processing strategy, the strategy was overridden to create the test programs.

All testing took place at the Washington University School of Medicine Department of Otolaryngology. Testing was completed with the participant seated in a double-walled sound-treated booth (8'3" x 8'11"). The R-Space loudspeakers were positioned in a circular pattern around the seated participant, with the face of each loudspeaker directed toward the center. The loudspeakers were placed at a distance of 24 inches from the participant, equally spaced in increments of 45º around the listener. See Figure 1 for a schematic diagram of the loudspeaker arrangement. Each loudspeaker was 44 inches above the ground, approximately at ear level for a seated adult.

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 run R-Space. The R-Space configuration is 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 is sent to four amplifiers (ART SLA-1, two-channel stereo linear power amp with 100 watts per channel) and then to eight loudspeakers (Boston Acoustic CR67) set in a circular array, 45º apart.

A Dell personal computer with a 24-bit studio sound card, a power amplifier, and a Urei 809A time align studio monitor loudspeaker was utilized to present CNC words in the soundfield. Participants were seated one meter from the loudspeaker at 0º azimuth.

Speech Test Materials

The R-Space noise was recorded in a busy restaurant that was to be later simulated in the test booth for more accurate evaluations of real-world directional benefit. 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 an equal 45º increments around his head (Compton-Conley et al., 2004).

The Hearing in Noise Test (HINT) sentences (Nilsson et al., 1994) were used to measure speech recognition in the R-Space noise. The HINT sentences consist of 25 recorded, phonetically balanced lists of 10 sentences each. The lists were written in American English and produced by a male speaker. Paired HINT lists numbers 1and 2 through 23 and 24 were used in this study.

Consonant-nucleus-consonant (CNC) words were used to measure speech recognition in quiet. The CNC word test consists of 10 lists of 50 words each (Peterson & Lehiste, 1962). CNC list numbers 1 through 10 (excluding lists 7 and 8) were used in this study.

Calibration

Calibration measurements were made using a Bruel and Kjaer 2230 sound level meter with a 1/3rd octave band filter attachment type 1625. An RMS detector method was used with slow time weighting. The selected frequency weighting was a band-pass filter from 20 Hz – 20 kHz. The microphone of the sound level meter was positioned facing upwards at a 90º angle relative to all eight loudspeakers to equally weigh the sound source from all loudspeakers.

Soundfield SPL measurements were previously made with 0 dB attenuation set within the MOTU Digital Performer 5 software and with the gains of the power amplifiers fixed. Individual sentence levels were measured for each pair of HINT lists (20 sentences total) and averaged to create a list level to be used for calibration. Next, the average output level was measured for the segment of restaurant noise during which each sentences are played. These measures were made to determine the dB SPL level with 0 dB attenuation for sentence lists and noise level per list so that correction factors may be applied to produce the desired presentation level of 70 dB SPL and accurate SNR levels during testing. The average output for each list of sentences and the corresponding restaurant noise section was verified separately prior to the start of this study to confirm that the appropriate SNR was obtained in the soundfield.

Word Recognition Testing

Additional testing was completed for the 25 participants utilizing a Freedom processor to evaluate conversion of Freedom processor programs to CP810 processor programs. CNC words were presented at 60 dB SPL in quiet at 0º azimuth with the participant's own processor and then with the CP810. Speech recognition performance was measured using two CNC word lists of 50 words per list for each condition. Processor order was counter-balanced and lists were randomly assigned.

R-Space Sentence Recognition Testing

Participants responded to HINT sentences presented from a loudspeaker at 0º azimuth with the R-Space noise presented from all eight loudspeakers at 70 dB SPL. A reception threshold for sentences (RTS) was obtained for each HINT list using an adaptive procedure in +2 dB step sizes for each processing option. A list of twenty sentences was presented for each of the different processing options. A hypothetical 21st trial was added in which the presentation level was predicted based on the response of the last sentence. The first four sentences are for acclimatization purposes and are not included in the calculation of the final RTS score. The last 17 presentation levels for trials 5 through 21 were averaged to represent an RTS score, in dB SNR, for each processing condition. The non-test ear was muffed and plugged when hearing thresholds were 60 dB HL or better.

Eight processing options were evaluated including Beam-only, Beam+ASC, Beam+ADRO, Beam+ASC +ADRO, Zoom-only, Zoom+ASC, Zoom+ADRO, and Zoom+ASC+ADRO. Processing option and list order was randomly assigned. Test session duration ranged from approximately 60 to 90 minutes and participants were offered a 5- to10-minute break to avoid fatigue during testing.

RESULTS

Statistical Analysis

Statistical analysis was completed to compare demographic and audiologic variables between the 25 Freedom recipients and the 7 CP810 recipients to determine if significant ($p \leq 0.05$) differences existed between processor groups. The variables of interest included the implanted ear (or test ear for bilateral participants), unilateral vs. bilateral CI use, age at testing, age at initial stimulation, years of implant use, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation. Each variable was compared between the CP810 and Freedom recipients using Fisher's exact test for categorical variables, unpaired t-tests for continuous variables, or Wilcoxon's test for instances in which assumption violations were detected. For the 25 Freedom recipients, paired t-tests were performed to compare CNC speech recognition in quiet with the CP810 and Freedom processors.

A one-way repeated measures analysis of variance (ANOVA) was used to analyze RTS scores among the eight processing options. Pairwise comparisons between select processing options were performed using statistical contrasts within the ANOVA model. These comparisons were determined *a priori* based on their potential for clinical relevance. These included comparisons between the four processing options utilizing Beam ("Beam group") and between those utilizing Zoom ("Zoom group"). Pairwise comparisons were also examined between the Beam and Zoom groups. These included Beam-only vs. Zoom-only, Beam+ASC vs. Zoom+ASC, Beam+ADRO vs. Zoom+ADRO, Beam+ASC+ADRO vs. Zoom+ASC+ADRO, and the entire Beam group vs. the entire Zoom group.

To provide additional information regarding performance differences between processing options, exploratory analyses were performed to examine potential demographic and audiologic variables. The variables of interest included participant age at testing, years of implant use, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation. Each covariate was entered into a repeated measures ANOVA, one variable per model, to evaluate whether the covariate was significant. In addition, the potential covariates were examined as bivariate variables to explore the interaction between the variable and performance across processing options. The continuous variables (age at testing, years of CI use, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation) were divided by the median and performance variability across processing strategies was compared for participants above and below the median. The ear of implantation (or ear randomly chosen for bilateral participants), was divided categorically.

Pearson product-moment correlation tests were used to explore associations across processing options, within the Beam and Zoom groups separately. Associations were further analyzed with correlations between participant demographic and audiologic characteristics and processing options. Correlation coefficients were computed for each variable and processing option. In addition, participant characteristics were investigated to determine if any impacted the performance differences between processing options. These associations were examined within the Beam and Zoom groups separately. All data analysis was generated using SAS software, version 9.2 of the SAS System for Linus (SAS Institute Inc., Cary, NC, USA).

Comparison of Freedom and CP810 Recipients

Statistical analysis of the Freedom and CP810 recipients was completed to determine if significant differences existed between processor groups. The variables examined included the implanted ear, unilateral vs. bilateral CI use, age at testing, age at initial stimulation, years of implant use, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation. Years of CI use was found to be significantly different between the CP810 and Freedom recipients ($W=28$; $p<0.0001$). The Freedom recipients had more CI experience with an average of 3.7 years, compared to an average of 1.0 years for the CP810 recipients. This was an expected finding because the CP810 is the most recently released processor that newly implanted patients received. No significant differences between processor groups were found for any of the other variables of interest.

Table 3 lists individual and mean CNC word and phoneme scores for the 25 Freedom recipients. This testing was done with the participants utilizing a Freedom processor in everyday life to evaluate conversion of the participant's Freedom processor program to a CP810 processor program. Word recognition performance ranged from 27% to 90% correct, with mean scores of 62.8% and 64.8% for the Freedom and CP810 processors, respectively. Mean word and phoneme scores are shown in Figures 2 and 3. No statistically significant difference was found for CNC word recognition performance between the Freedom and CP810 processors. Phoneme recognition performance ranged from 50% to 96% correct. Phoneme scores were significantly higher ($t=-2.28$; $p = 0.03$) when participants were tested with the CP810 versus the Freedom, with mean scores of 81.6% and 79.8% correct, respectively.

R-Space Sentence Recognition Testing

The mean RTS scores and standard deviations for each of the four processing options utilizing Beam (Beam-only, Beam+ASC, Beam+ADRO, Beam+ASC+ADRO) are shown in Figure 3. A lower RTS indicates better speech recognition performance in noise. The best performance was found with Beam-only with a mean RTS of 6.2 dB. The mean RTS scores for Beam+ASC and Beam+ASC+ADRO were equivalent at 6.5 dB and 6.6 dB, respectively. Beam+ADRO resulted in the poorest performance with a mean RTS of 7.8 dB. ANOVA revealed a significant decrease in performance with Beam+ADRO compared to all other Beam processing options. Beam+ADRO had significantly higher mean RTS scores compared to Beam-only [$F(1,31)=12.55$; $p=.001$], Beam+ASC [$F(1,31)=9.87$; $p=.004$], and Beam+ASC+ADRO [$F(1,31)=5.23$; $p=.03$]. There was no statistical difference between Beam-only and Beam+ASC or Beam+ASC+ADRO. No significant difference was observed between Beam+ASC and Beam+ASC+ADRO. Fifteen individuals performed best with processing options utilizing Beam: 6 with Beam-only, 5 with Beam+ASC, 1 with Beam+ADRO, and 3 with Beam+ASC+ADRO. Figure 4 shows the percentage of participants who performed best with each of the Beam processing options.

Figure 5 shows the mean RTS scores and standard deviations for each of the Zoom processing options (Zoom-only, Zoom+ASC, Zoom+ADRO, Zoom+ASC+ADRO). Zoom+ASC resulted in the lowest, or best performance with a mean RTS of 5.7 dB. The poorest performance was observed with Zoom-only, with a mean RTS of 7.9 dB. Mean RTS scores for Zoom+ADRO and Zoom+ASC+ADRO were 7.6 dB and 6.9 dB, respectively. Zoom+ASC revealed significantly better mean RTS scores compared to Zoom-only [$F(1,31)=12.30$; $p=.001$] and Zoom+ADRO [$F(1,31)=8.03$; $p=.008$]. No significant difference was observed between Zoom+ASC and Zoom+ASC+ADRO, although better performance was observed with Zoom+ASC. Zoom+ASC+ADRO was not significantly better than Zoom+ADRO or Zoom-only. No significant difference was observed between Zoom+ADRO and Zoom-only. Seventeen individuals performed best with the Zoom processing options: 4 with Zoom-only, 10 with Zoom+ASC, 0 with Zoom+ADRO, and 3 with Zoom+ASC+ADRO. Figure 6 shows a breakdown of the percentage of participants who performed best with each of the Zoom processing options.

The differences in performance among the eight processing options are depicted in Figure 7. Pairwise comparisons between processing options in the Beam and Zoom groups showed a statistically significant difference in performance between Beam-only compared to Zoom-only [$F(1,31)=7.47$; $p=.01$], with mean RTS scores of 6.2 dB and 7.9 dB, respectively. No other significant differences in mean performance were found for any of the other between-group comparisons.

Figure 8 shows the performance for each of the eight processing options in order of lowest (best) to highest (poorest) mean RTS scores. Overall, the best performance was observed with Zoom+ASC, with a mean RTS of 5.73 dB. Zoom-only resulted in the poorest performance, with a mean RTS of 7.94 dB. A one-way repeated measures (ANOVA) revealed a significant effect of processing strategy [$F(7,31)=4.54$; $p= .0014$].

A large amount of individual variability in performance was found in this study. Figure 9 shows the individual best (lowest) RTS scores and the corresponding processing options for the participants in the current study. The largest percentage of participants (31%; n=10) performed best with Zoom+ASC, which also had the best mean RTS score. Interestingly, 13% (n=4) of the participants performed the best with Zoom-only, which had the poorest mean RTS score of the eight processing options. Three of these participants scored in the 75th percentile, meaning that 75% of the participants demonstrated poorer performance than these individuals.

Potential Moderators

Additional analysis was performed to determine if the results of the R-Space sentence recognition testing were moderated by other variables. Variables examined included implanted ear, age at testing, years of implant use, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation. Two analyses of the variables were completed. For covariates, each variable was entered separately into the ANOVA model, with one variable per model. This analysis showed no significant interactions for any of the variables examined, thus suggesting that these variables do not provide information regarding performance variability among processing options.

Bivariate analysis was then performed on ear of implantation and dichotomized covariates. No significant interaction was observed between performance differences across processing options for any of the variables. Thus, performance differences between processing options were the same for recipients implanted in the right ear and recipients implanted in the left ear. Performance differences were the same for recipients less than 71.5 years old and recipients 71.5 years or older. Performance differences were the same for recipients with less than 2.8 years of CI use and recipients with 2.8 or more years of CI use. Performance differences were the same for recipients with less than 31.5 years of hearing loss and recipients with 31.5 or more years of hearing loss prior to implantation. Performance differences were the same for recipients with less than 6 years of severe-to-profound hearing loss and recipients with 6 or more years of severe-to-profound hearing loss prior to implantation. Performance differences were the same for recipients with less than 18 years of hearing aid use and recipients with 18 or more years of hearing aid use prior to implantation.

Correlation Analysis

Correlational analysis was completed to examine individual trends in performance between processing options within the Beam and Zoom groups separately. The results of this analysis are shown in Table 4. Significant correlations were found across processing options utilizing Beam. Positive linear trends were observed which suggests that the processing options were highly associated. Significant correlations were also found for the Zoom group, with positive linear trends representing strong associations between processing options utilizing Zoom.

Associations were analyzed between participant characteristics and performance within each processing option. Variables examined included implanted ear, age at testing, years of implant use, years of hearing loss, years of severe-to-profound hearing loss, and years of hearing aid use prior to implantation. Correlation coefficients were completed for each variable, within each processing option. The only significant positive correlation was between Zoom+ASC+ADRO and the number of years of severe-to-profound hearing loss prior to implantation ($r=.38$; $p=.03$). However, caution must be used in evaluating the significance of this individual correlation. A large number of statistical tests were conducted, thus increasing the likelihood that any one of these correlations is significant by chance alone. No other significant correlations were found between any of the variables and processing options.

Further analysis was performed to examine associations between participant characteristics and performance differences between processing options. This analysis was performed within the Beam and Zoom groups separately. No statistically significant correlations were found between any of the participant characteristics and performance variability between processing options for either processing group.

DISCUSSION

Many CI recipients achieve high levels of speech recognition in quiet but experience decreased performance in background noise. Since many CI recipients report difficulty understanding speech in noise, a main goal for cochlear implant manufacturers is to improve technology to address this issue. The purpose of the current study was to evaluate the speech processing options currently available in the CP810 sound processor using the R-Space speaker setup to determine which processing option(s) performs best in R-Space background noise.

Eight processing options were evaluated including Beam-only, Beam+ASC, Beam+ADRO, Beam+ASC+ADRO, Zoom-only, Zoom+ASC, Zoom+ADRO, and Zoom+ASC+ADRO. Mean RTS scores in order of best to poorest performance were as follows: Zoom+ASC (5.7 dB), Beam-only (6.2 dB), Beam+ASC (6.5 dB), Beam+ASC+ADRO (6.6 dB), Zoom+ASC+ADRO (6.9 dB), Zoom+ADRO (7.6 dB), Beam+ADRO (7.8 dB), and Zoom-only (7.9 dB).

Performance with Beam-only was significantly better than Zoom-only. One reason for this finding may be that Beam utilizes adaptive directionality and is therefore able to take advantage of the instantaneous loudness variations of the R-Space noise between loudspeakers. See Figure 1 for a schematic diagram of the R-Space array, with loudspeakers numbered 1-8. With a fixed null at ±120º, Zoom will cancel noise most effectively between loudspeakers 3 & 4 or 6 & 7. Beam, however, utilizes adaptive directionality to reduce the most intense noise source arriving between 90º and 270º azimuth (loudspeakers 3-7). Beam-only directional processing may have done a better job of limiting R-Space background noise than Zoom-only because it chooses its null from a wider range of azimuths. Also, the second adaptive noise cancellation stage utilized by Beam, but not by Zoom, was likely very beneficial in the R-Space. The speech reference in Zoom and Beam is created using a forward facing directional pattern. Since the R-Space noise is presented from all azimuths around the listener, including the front, the speech reference most likely still contained noise. The adaptive noise cancellation stage is designed to reduce the remaining noise in the speech reference. Therefore, it is likely that Beam was able to more accurately separate the speech from the noise in the speech signal.

The best performance was found with Zoom+ASC with an RTS of 5.7 dB, while Zoom-only yielded the worst performance with an RTS of 7.9 dB. The difference between the two processing options was statistically significant. This suggests that the fixed directional mode utilized by Zoom needs ASC to further filter out the background noise. This finding was not observed with Beam. No significant difference in performance existed between Beam-only and Beam+ASC, with RTS scores of 6.2 and 6.5 dB, respectively. In contrast to Zoom, where ASC's noise processing significantly improved performance, the addition of ASC's noise processing to Beam resulted in almost no change in performance compared to Beam-only.

Since it appears that ASC is mainly responsible for the improved performance observed with Zoom+ASC, it would have been interesting to evaluate ASC-only. Also of interest would have been a comparison between ASC-only and Beam-only, since ASC is clearly providing benefit with Zoom, but not with Beam. While this study did not examine ASC-only processing, a similar study by Brockmeyer and Potts (2011) evaluated ASC and Beam in 70 dB R-Space noise for Freedom recipients. ASC yielded a slightly better mean RTS score of 9.7 dB, compared to Beam with an RTS of 11.4 dB. This difference was not statistically significant, suggesting that relatively similar performance can be expected with either processing option.

The current study found a mean RTS of 6.2 dB with Beam-only processing in the CP810, which is a 5.2 dB improvement in performance compared to the results obtained by Brockmeyer and Potts (2011) with the Freedom Beam. This difference is in agreement with Cochlear's reported 5 dB improvement with the dual omnidirectional microphones utilized in the CP810 Beam (Cochlear Limited, 2010a). Conversely, Gifford and Revit (2010) reported a mean RTS of 7.3 dB with Beam+ASC+ADRO in the Freedom processor, which is similar to the 6.6 dB mean RTS for Beam+ASC+ADRO with the CP810 processor in the current study. Gifford and Revit

(2010) did not test Beam-only and therefore a direct comparison cannot be made between studies based strictly on the performance of Beam. This may suggest that the combination of processing options may provide benefit, regardless of the type of microphone configuration utilized in the processor. These findings question the definite improvement suggested for the CP810 processor. However, the results of the present study are specific to the speech recognition abilities of the individuals in this study, so performance with the processing options cannot be directly compared between these studies and the current study.

An overall trend of decreased performance was observed with the addition of ADRO. Zoom+ADRO and Beam+ADRO yielded the poorest performance, aside from Zoom-only (7.9 dB), with RTS scores of 7.6 and 7.8 dB, respectively. Performance decreased significantly with Zoom+ADRO compared to Zoom+ASC, with an RTS score decrease of 1.9 dB. Beam+ADRO yielded an RTS of 7.8 dB, which was significantly poorer than all other Beam conditions by 1.2 to 1.6 dB. It is possible that the performance found with ADRO is related to the programming protocol followed by the Washington University School of Medicine Department of Otolaryngology. All CI recipients are programmed according to a detailed protocol (Skinner et al., 1995; Sun et al., 1998; Skinner et al., 1999, Holden et al., 2002; Skinner et al., 2002). Threshold (T) level stimulation is set at or above the level at which the individual correctly counts sets of biphasic pulses. Maximum comfort (C) levels are programmed at a level judged as loud, but comfortable. Thus, the individual's electrical dynamic range is carefully measured and T and C levels are set to appropriately stimulate within this range.

How the T and C levels are programmed may affect the functioning of ADRO. ADRO functions to alter the gain of the acoustic input signal to place the signal optimally in the listener's dynamic range. The gain is increased or decreased based on several rules for audibility and loudness comfort. ADRO applies compression to high level noise to lower the input to a comfortable level within the listener's dynamic range. Although this may be beneficial when the loud input consists only of noise, complex interactions may occur when loud speech is presented at the same time as loud noise (James et al., 2002). In the current study loud speech in high-level noise was used. It is, therefore, possible that ADRO over-compressed the loud input to adhere to its "comfort" or "background noise" rules. In this case, the loud speech was lowered to an input level that would be mapped lower in the listener's dynamic range. The C level stimulation for the participants in this study was programmed on the louder side of comfortable, so ADRO's gain reduction may have compressed the speech to a greater degree than for a recipient with C levels programmed differently. ADRO may have improved comfort for the participants at the expense of audibility for the participants in the current study. Additionally, ADRO's audibility rule increases gain for soft sounds in an attempt to improve speech recognition. The participants in this study were programmed with threshold stimulation levels set above or well above detection levels. The advantage of this approach is that the listeners have greater access to soft speech. The disadvantage is that it may increase audibility for all low level input, including ambient noise. For the participants in this study, ADRO's gain rule may have increased the ambient noise to a level that was mapped higher in the recipient's dynamic range compared to a recipient with lower set T levels. It is possible this may also have had a detrimental effect on performance. All in all, ADRO's gain adjustments may not have resulted in optimal placement of the signal in the listener's dynamic range due to the interaction of the loud speech and noise utilized in this study and the way in which T and C level stimulation was programmed for the participants.

A decrease in performance was found with Zoom+ASC+ADRO compared to Zoom+ASC, although this difference was not statistically significant. This suggests that with all three input processing strategies in operation, ASC is likely responsible for the majority of improvement in performance, regardless of the addition of ADRO. A decrease in performance was also observed with Beam+ASC+ADRO compared to Beam+ASC, although this difference was not statistically significant. In this case, either Beam or ASC may be the predominant contributor to performance benefit and the reason for the relative stability in performance despite the use of ADRO.

Only some of the differences between processing options evaluated in this study were found to be statistically significant. The largest change in performance was found between Zoom+ASC and Zoom-only, with a significant difference of 2.2 dB. Other significant differences were identified with comparisons involving Beam+ADRO and Zoom+ADRO. While other comparisons were not found to be statistically significant, they may be clinically relevant. The performance differences across the Beam processing options, excluding Beam+ADRO, were relatively small, with an RTS difference of 0.4 dB from the best to poorest performance. However, performance with Zoom+ASC was 1.2 dB better than performance with Zoom+ASC+ADRO. Soli and Nilsson (1994) reported an approximate 10% improvement in sentence recognition in noise for every 1 dB SNR increase in HINT scores. Therefore, the addition of ADRO to Zoom+ASC processing could potentially decrease speech recognition by 12%.

Several variables must be considered when quantifying improvements in speech recognition performance with the processing options. The findings of this study are specific to the type, level, and configuration of the test materials utilized. The R-Space noise utilized in this study was specific to a particular restaurant at which the noise recordings were made. Thus, the findings of this study cannot be generalized to all noisy environments. Also, HINT sentences were presented from a loudspeaker at 0º azimuth and noise was presented from eight loudspeakers arranged in a circular pattern around the participant at 70 dB SPL. The noise was presented from all eight loudspeakers, including the loudspeaker at 0º azimuth. Compton and Conley (2004) reported significantly better speech recognition performance for adult hearing aid listeners when noise was presented at 180º directly behind the listener compared to the diffuse R-Space condition. The RTS scores in the current study may have improved if the participants were tested in an environment where noise was not presented in front of the listener. Also, larger differences between processing options may have occurred with the use of more discrete noise sources. Ricketts and Henry (2002) report that the advantages of adaptive directionality in hearing aids are mainly expected in environments where the noise source was relatively discrete, but the differences between adaptive and fixed directional processing were less clear in diffuse noise.

In addition, the results of the current study are specific to the high level of noise that was chosen. Previous research has shown that louder noise results in higher (poorer) SRT scores. The magnitude of the decrease is different between processing options. Brockmeyer and Potts (2011) reported slightly poorer performance with ASC and significantly poorer performance with Beam in 70 dB compared to 60dB R-Space noise. The difference in performance at louder noise levels may simply be attributed to the increased difficulty encountered in louder environments or to the ability of the processing options to handle the increased noise level. Beam, when active, will constantly adapt to minimize the output of the loudest noise source. How effectively Beam performs this task is dependent on the type of noise and the signal-to-noise ratio. For steady state noise, Beam is expected to perform well to a SNR of approximately -5 dB (Cochlear Support, 2011). The SNR at which Beam is no longer effective is unknown for babble noise, but would be expected to be higher than steady state noise.

Another factor that must be considered is the level at which ASC processing is activated. ASC monitors the noise floor by analyzing troughs in envelope of the input signal. During breaks in speech, the level of the noise floor is estimated. If the noise floor is above 57 dB, ASC reduces the microphone sensitivity so that the peaks of speech exceed the long-term average noise spectrum by at least 15 dB (Wolfe, 2009). Depending on the level of the signal and the noise, ASC may or may or may not be active. The current study utilized a 70 dB SPL noise level to increase the likelihood that ASC was active during the testing and the use of a lower noise level may have yielded different results.

Finally, the interaction between processing options and the participant's programmed stimulation levels is unknown. All participants in this study were programmed using the same detailed protocol. Threshold levels were set above (or well above) first detection and C levels were set at a loud, but comfortable level. The processing options are designed to optimize speech recognition performance by modifying the amount of stimulation received by the recipient. A different programming protocol may have elicited different results due to the different interactions between the processing options and the recipient's programmed stimulation levels.

The results of this study may influence CI programming decisions by providing additional suggestions, beyond Cochlear's recommended default settings, for which noise processing programs available in the CP810 may be most effective in optimizing speech understanding in noise. However, there was a large amount of individual variability in performance found in this study. This suggests that functional benefit obtained with the processing options varies between CI recipients. In addition, subjective preference was not examined in this study but may be inconsistent with performance measured in the clinic. It is important to use different processing options on an individual basis to determine which option achieves optimal speech understanding and subjective preference.

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Table 1: Individual demographic and audiologic information. Group means and standard deviations are listed at the bottom of the table.

Participant	Gender	Age	Implanted Ear	Years of HL	Years of Severe to Profound HL	Years of HA Use	Etiology
1	F	79	R	11	1	7	Unknown
2	M	66	L	54	2	5	Measles
3	F	44	R	12	5	5	Unknown
4	M	63	L	22	1	17	Noise exposure
5	F	47	R	18	3	12	Unknown
6	F	56	L	55	14	35	Usher's type II
7	M	78	R	32	6	18	Noise
8	F	82	L	51	10	29	Unknown
9	F	83	R	48	30	35	Genetic
10	M	79	R	26	5	6	Unknown
11	F	55	R	9	3	6	Genetic
12	F	83	L	43	14	17	Otosclerosis
13	F	72	L	17	6	0	Meniere's Disease
14	F	36	R	31	31	29.5	Unknown
15	M	52	R	45	8	44	Maternal Rubella
16	F	42	R	32	5	0	Unknown
17	M	51	L	44	38	42	Unknown
18	M	79	L	42	5	22	Otosclerosis
19	M	75	R	20	3	18	Noise
20	M	79	R	57	4	41	Noise
21	F	48	L	26	1	25	Unknown
22	M	72	L	24	22	1	Ototoxicity
23	F	73	L	18	2	15	Autoimmune disease
24	F	78	L	31	13	28	Unknown
25	M	79	L	48	1	48	Meniere's Disease
26	M	66	R	20	4	0	Unknown
27	M	70	L	49	15	30	Unknown
28	M	75	L	51	11	18	Noise exposure
29	M	92	L	26	10	25	Noise exposure
30	M	59	L	15	10	11	Unknown
31	M	51	R	45	45	39	Meningitis
32	M	63	L	40	8	1	Unknown
Mean		66.9		33.2	10.5	19.7	
SD		14		15	11	15	

Table 2: Individual information related to CI use. Group mean and standard deviation for years of CI use is listed at the bottom of the table.

Participant	Processor	Strategy	Rate (Hz)	Maxima	Years Of Implant Use	Preferred Everyday Preprocessing
1	Freedom	ACE	1200	10	6.9	ASC+ADRO
2	Freedom	ACE	1800	8	3.5	ADRO
3	CP810	ACE	1800	8	0.8	Beam+ASC+ADRO
4	Freedom	ACE	1800	8	3.3	ASC
5	Freedom	ACE	900	8	5	None
6	CP810	ACE	1800	8	0.8	None
7	Freedom	ACE	1800	8	4	None
8	Freedom	ACE	900	10	3.7	None
9	Freedom	ACE	1200	10	4.2	None
10	Freedom	ACE	2400	10	4.1	Beam
11	Freedom	ACE	900	8	1.8	ASC+ADRO
12	Freedom	ACE	1200	10	3.9	None
13	Freedom	ACE	1200	8	4.2	ADRO
14	Freedom	ACE	1200	12	5	ASC
15	Freedom	ACE	2400	10	4.8	None
16	Freedom	ACE	1800	10	5.3	None
17	Freedom	ACE	1800	8	4.9	None
18	Freedom	ACE	1200	8	2.2	Beam+ASC+ADRO
19	Freedom	ACE	720	8	2.1	Beam+ASC
20	Freedom	ACE	1200	8	2.8	ADRO
21	Freedom	ACE	1800	8	2.8	ADRO
22	Freedom	ACE	1200	10	2.8	ADRO
23	Freedom	ACE	1200	8	2	ADRO
24	Freedom	ACE	1800	8	2	ASC+ADRO
25	Freedom	ACE	1200	8	1.9	ADRO
26	CP810	ACE	900	8	1	ADRO
27	Freedom	ACE	900	10	2.6	ADRO
28	CP810	ACE	900	10	1.3	ADRO
29	CP810	ACE	900	10	1.4	ADRO
30	CP810	ACE	1200	10	0.7	ADRO
31	Freedom	ACE	1200	8	5.8	None
32	CP810	ACE	500	8	1.2	ADRO
Mean					3.1	
SD					1.6	

Figure 1: Schematic diagram of the R-Space loudspeaker arrangement. Figure taken from Compton-Conley et al. (2004) with the addition of loudspeaker numbers.

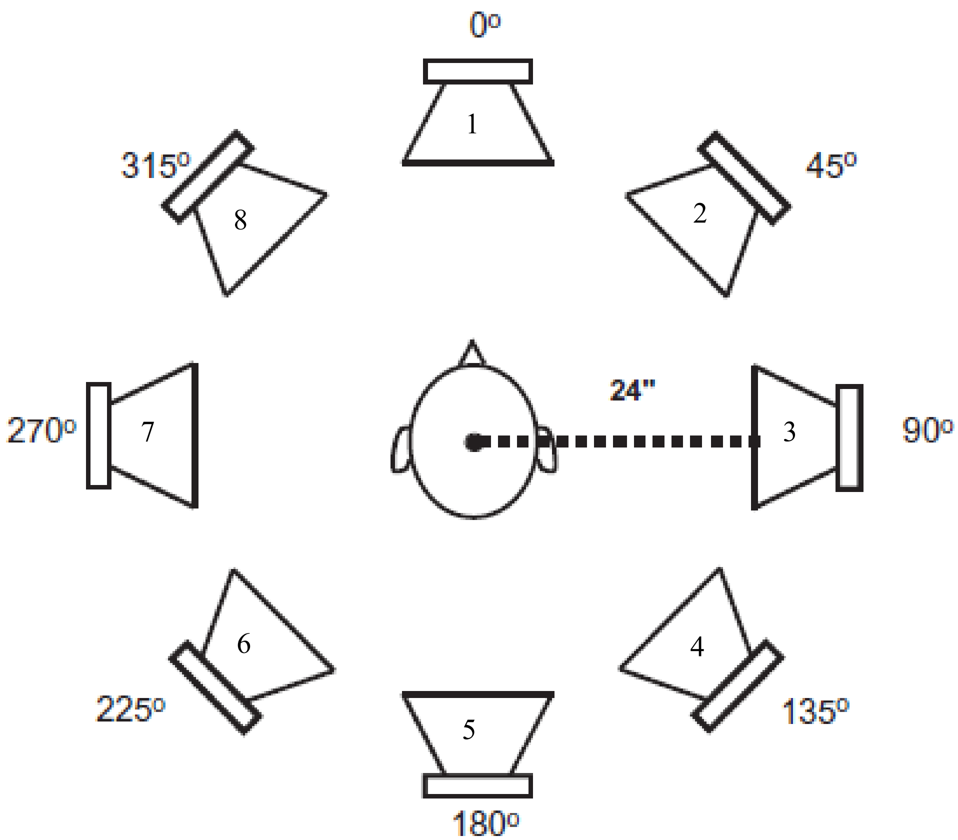

Table 3: Individual CNC word and phoneme scores (% correct) for Freedom users. Participants were tested using their own processor and a CP810 processor programmed with equivalent settings. Group means and standard deviations are listed at the bottom of the table.

Participant	Freedom Words	CP810 Words	Freedom Phonemes	CP810 Phonemes
1	85	85	93	91
2	89	86	95	94
4	83	83	93	93
5	59	61	78	82
7	55	57	72	77
8	70	71	86	85
9	85	90	95	96
10	27	29	54	65
11	86	77	96	89
12	77	85	89	94
13	53	63	75	80
14	47	50	73	77
15	68	60	85	80
16	85	80	94	93
17	56	59	79	78
18	64	69	82	85
19	45	51	70	74
20	57	53	78	76
21	85	88	93	96
22	52	49	71	70
23	28	31	50	53
24	73	81	87	91
25	47	57	68	77
27	29	35	55	61
31	66	71	84	87
Mean	62.8	64.8	79.8	81.8
SD	19.2	18	39.7	34

Figure 2: Mean CNC word and phoneme scores (% correct) for Freedom users. Error bars represent +1 standard deviation. The asterisks represent a significant difference (p≤0.05) between scores.

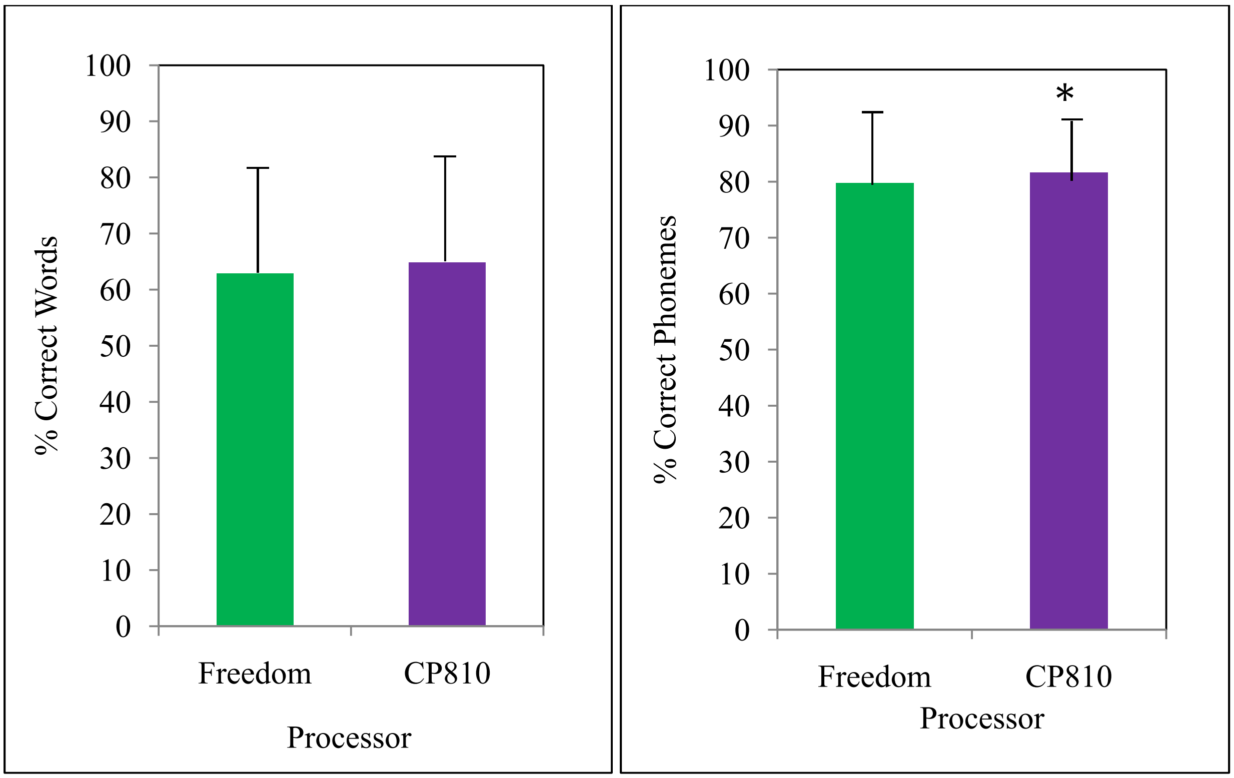

Figure 3: Mean RTS scores with Beam-only, Beam+ASC, Beam+ADRO, and Beam+ASC+ADRO processing options. Error bars represent +1 standard deviation. The asterisks represent a significant difference between processing options ($p \leq 0.05$).

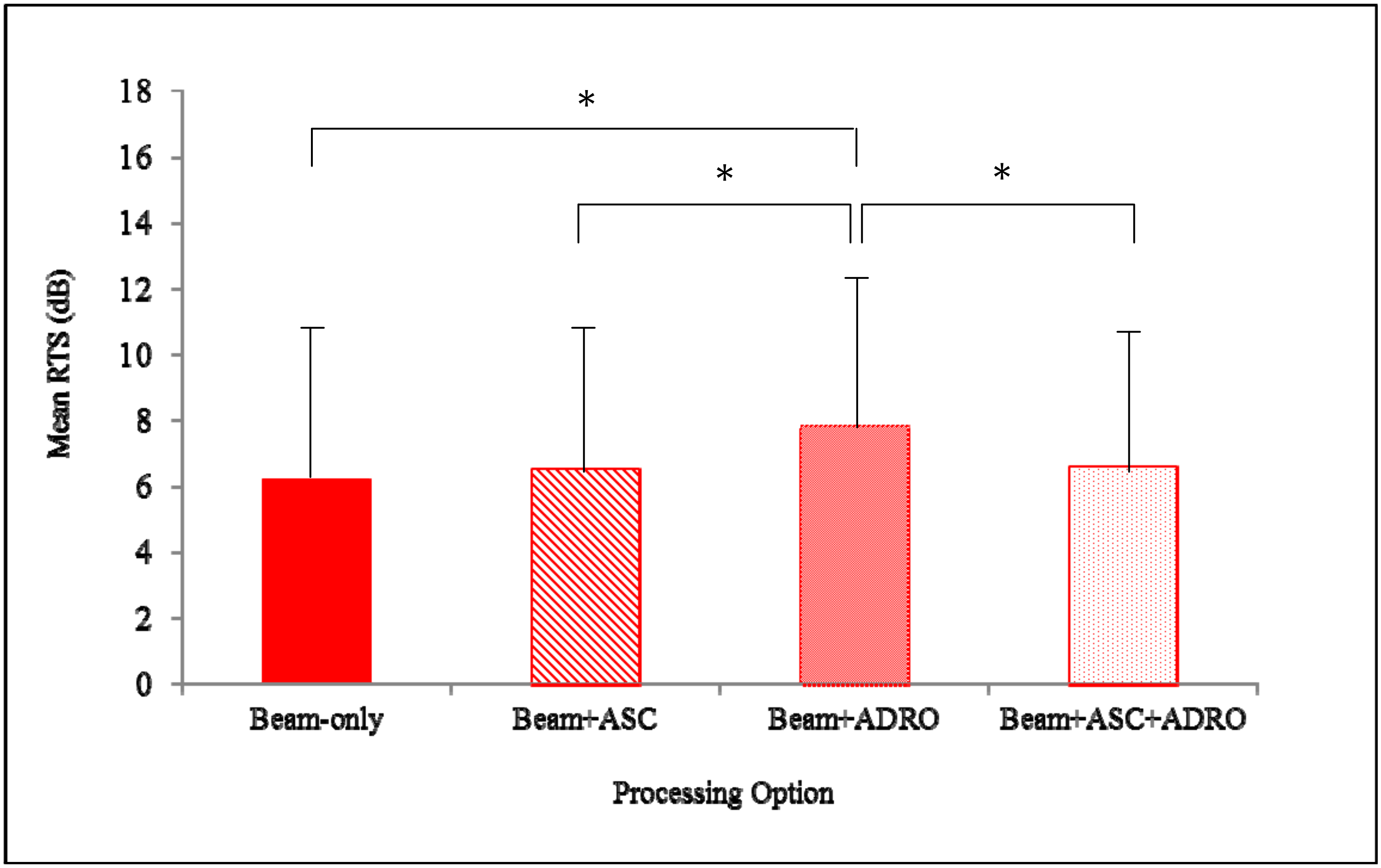

Figure 4: Breakdown of participants who performed best with Beam processing options. Percentages are relative to the 15 participants who performed best within the Beam group.

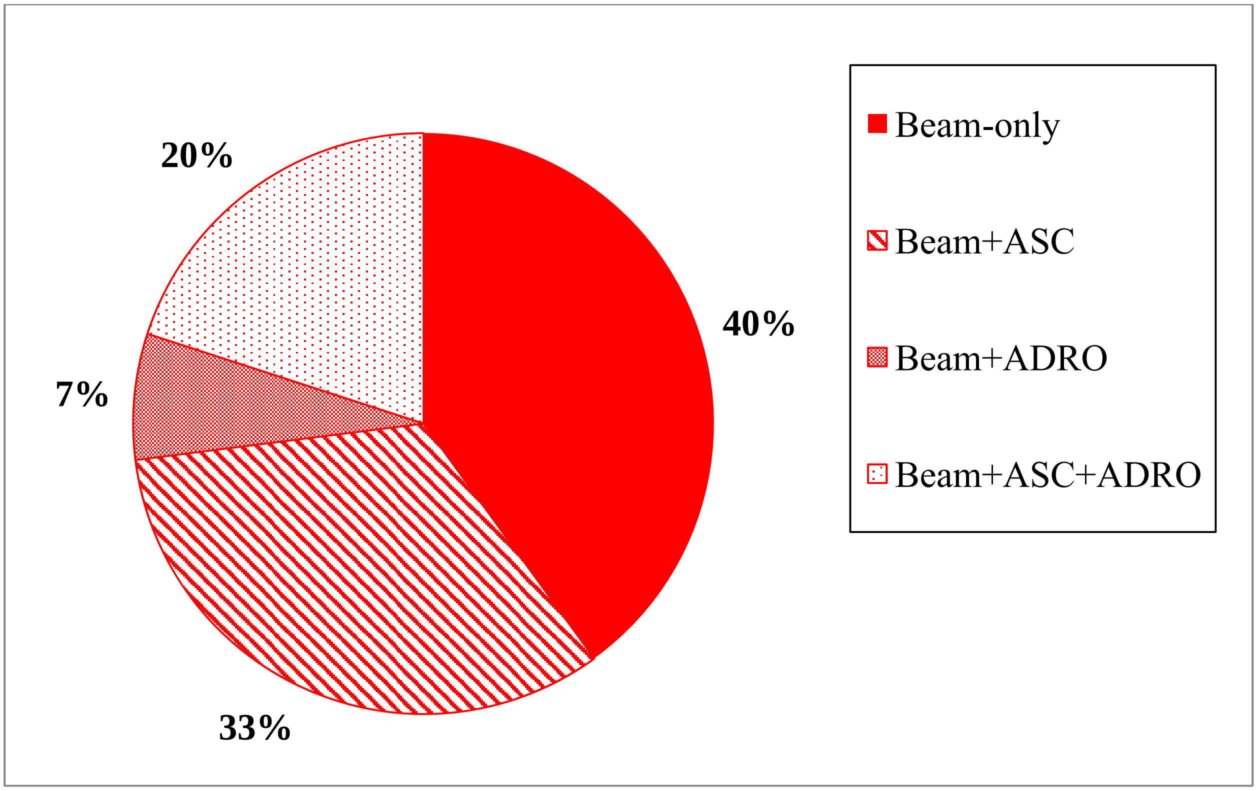

Figure 5: Mean RTS scores with Zoom-only, Zoom+ASC, Zoom+ADRO, and Zoom+ASC+ADRO processing options. Error bars represent +1 standard deviation. The asterisks represent a significant difference between processing options (p≤0.05).

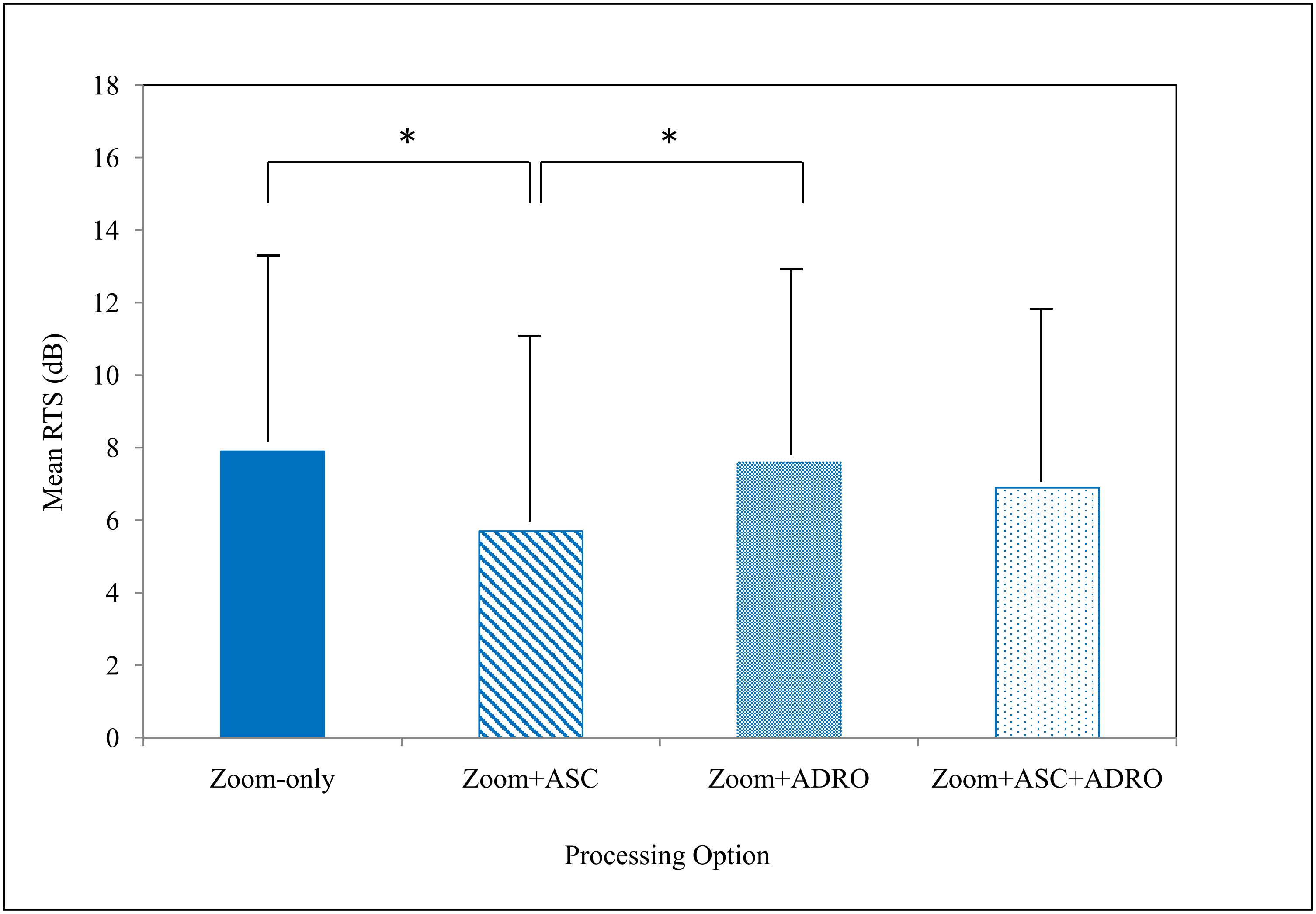

Figure 6: Breakdown of participants who performed best with Zoom processing options. Percentages are relative to the 17 participants who performed best within the Zoom group**.**

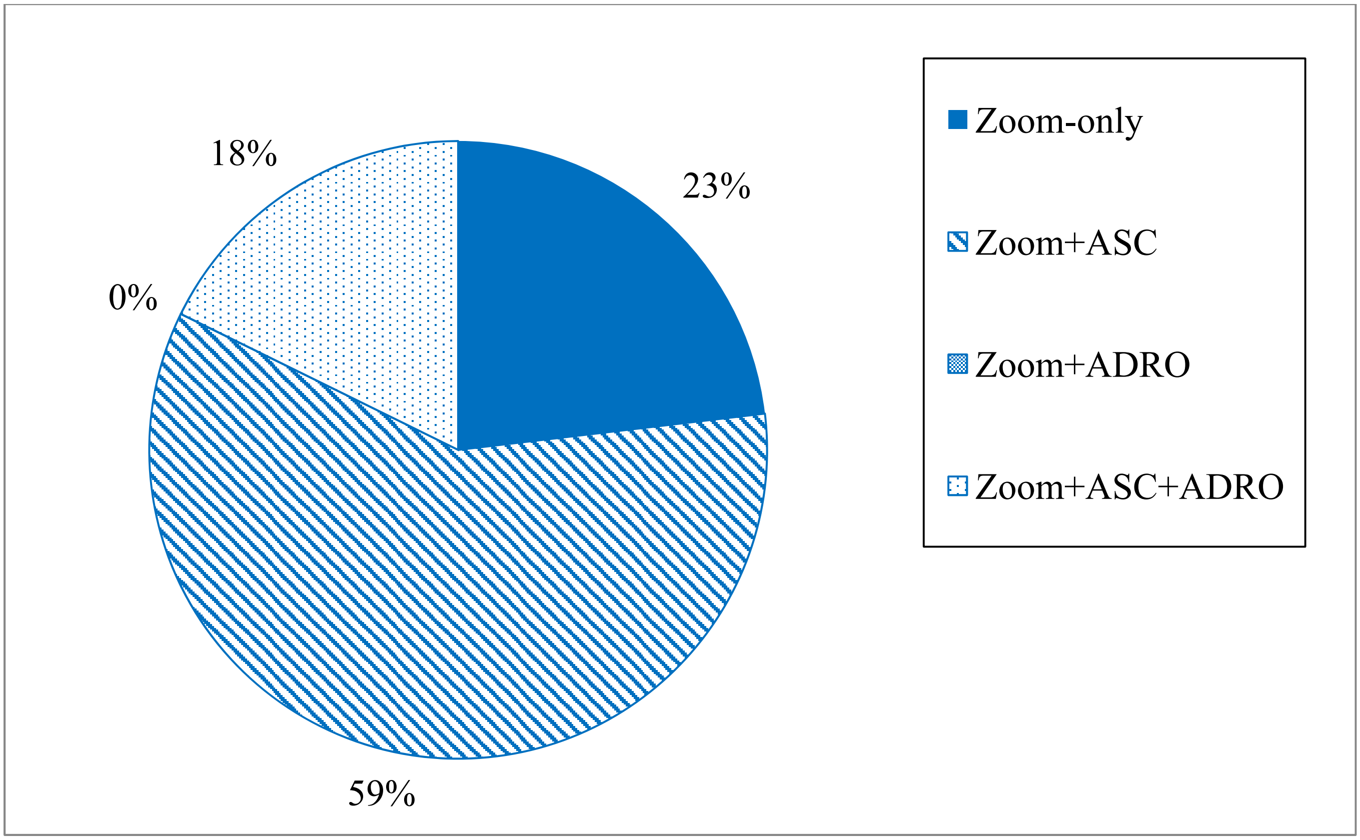

Figure 7: Mean RTS for participants with Beam-only, Zoom-only, Beam +ASC, Zoom+ASC, Beam+ADRO, Zoom+ADRO, Beam+ASC+ADRO, and Zoom+ASC+ADRO processing options. Error bars represent +1 standard deviation. The asterisks represent a significant difference between processing options (p≤0.05).

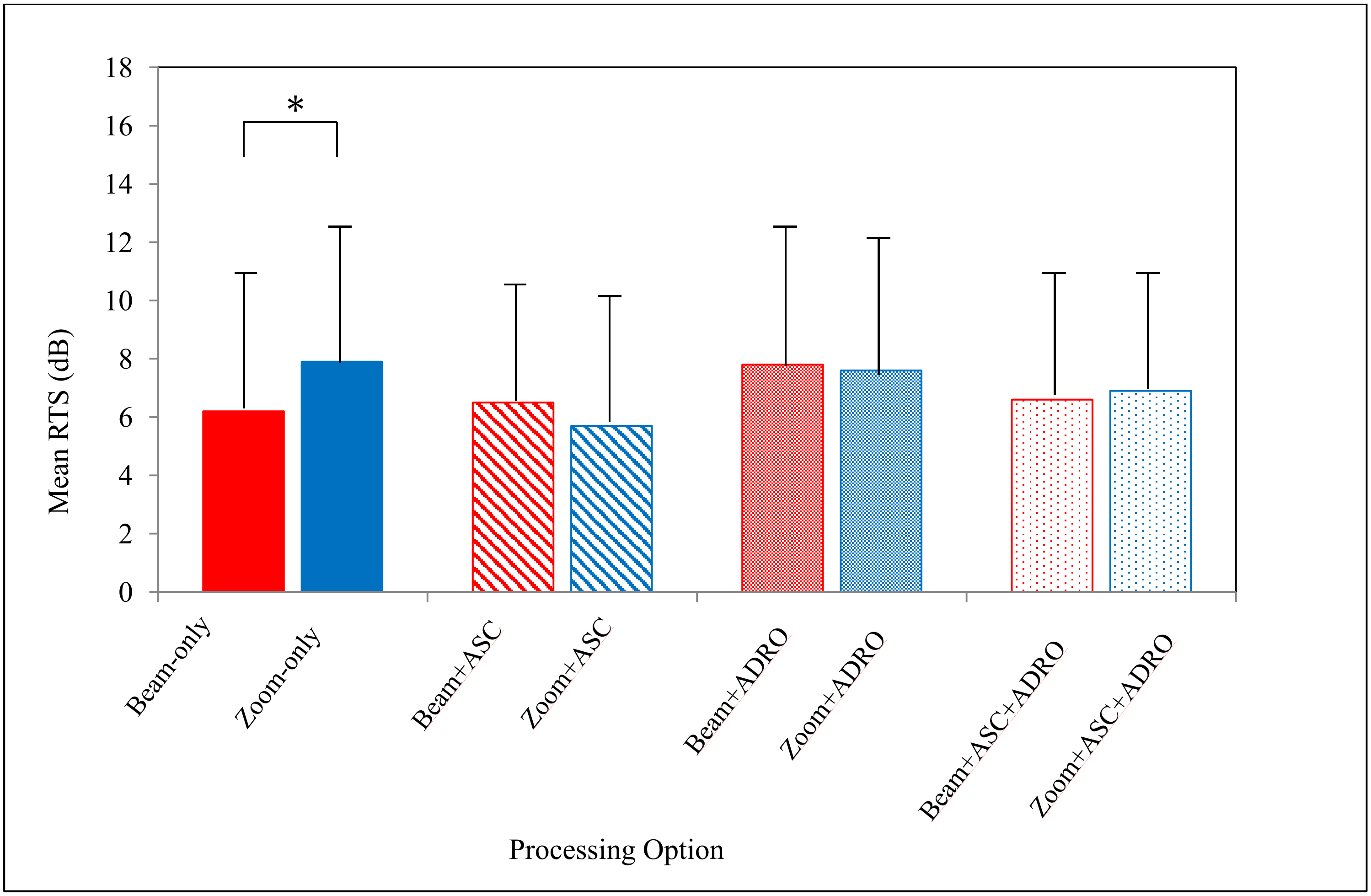

Figure 8: Mean RTS scores with each of the eight processing options. Performance is shown in order of lowest (best) to highest (poorest) RTS scores.

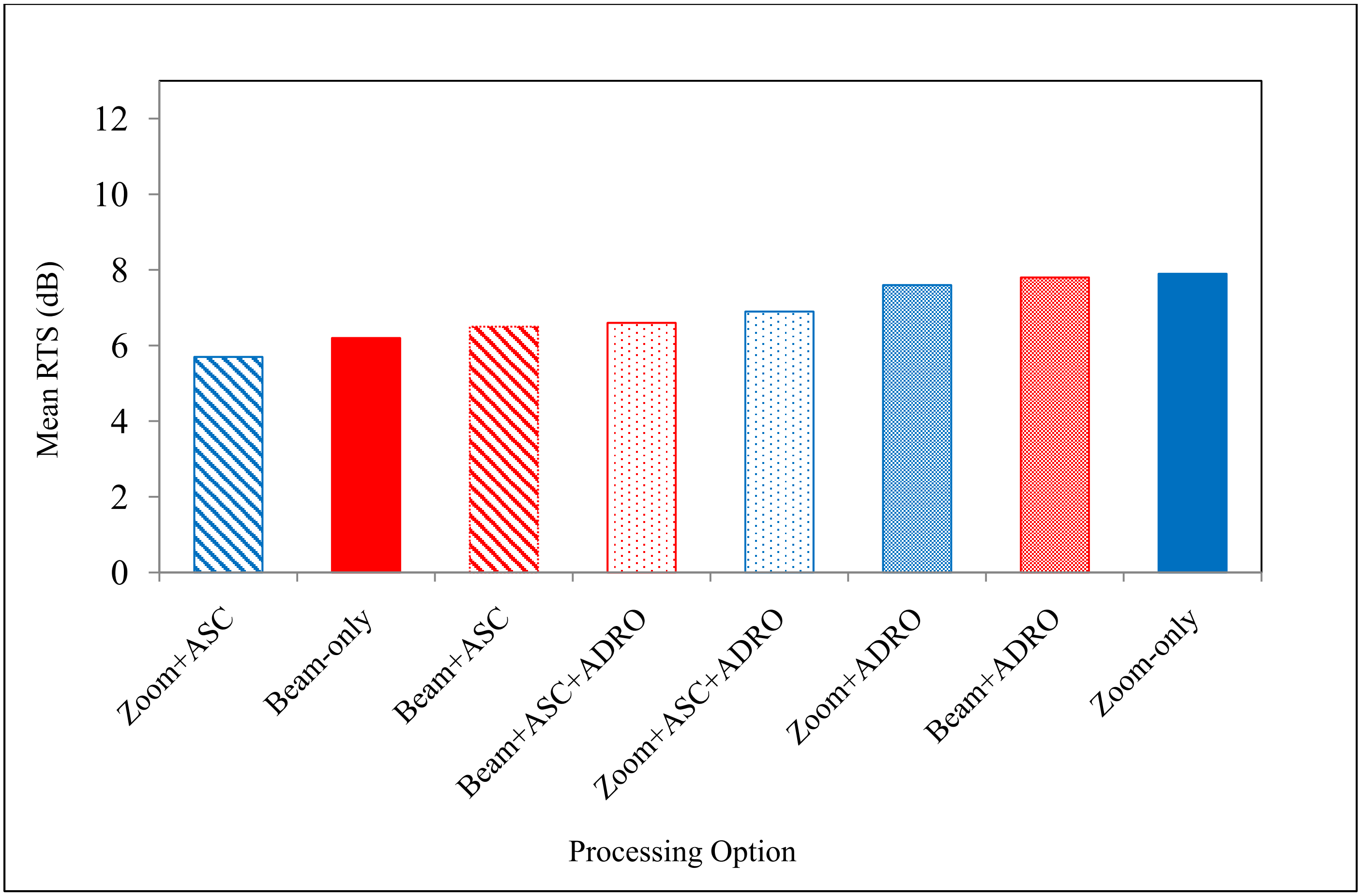

Figure 9: Individual participants' best (lowest) RTS scores and the corresponding processing options. Performance is shown in order of the lowest to highest best scores.

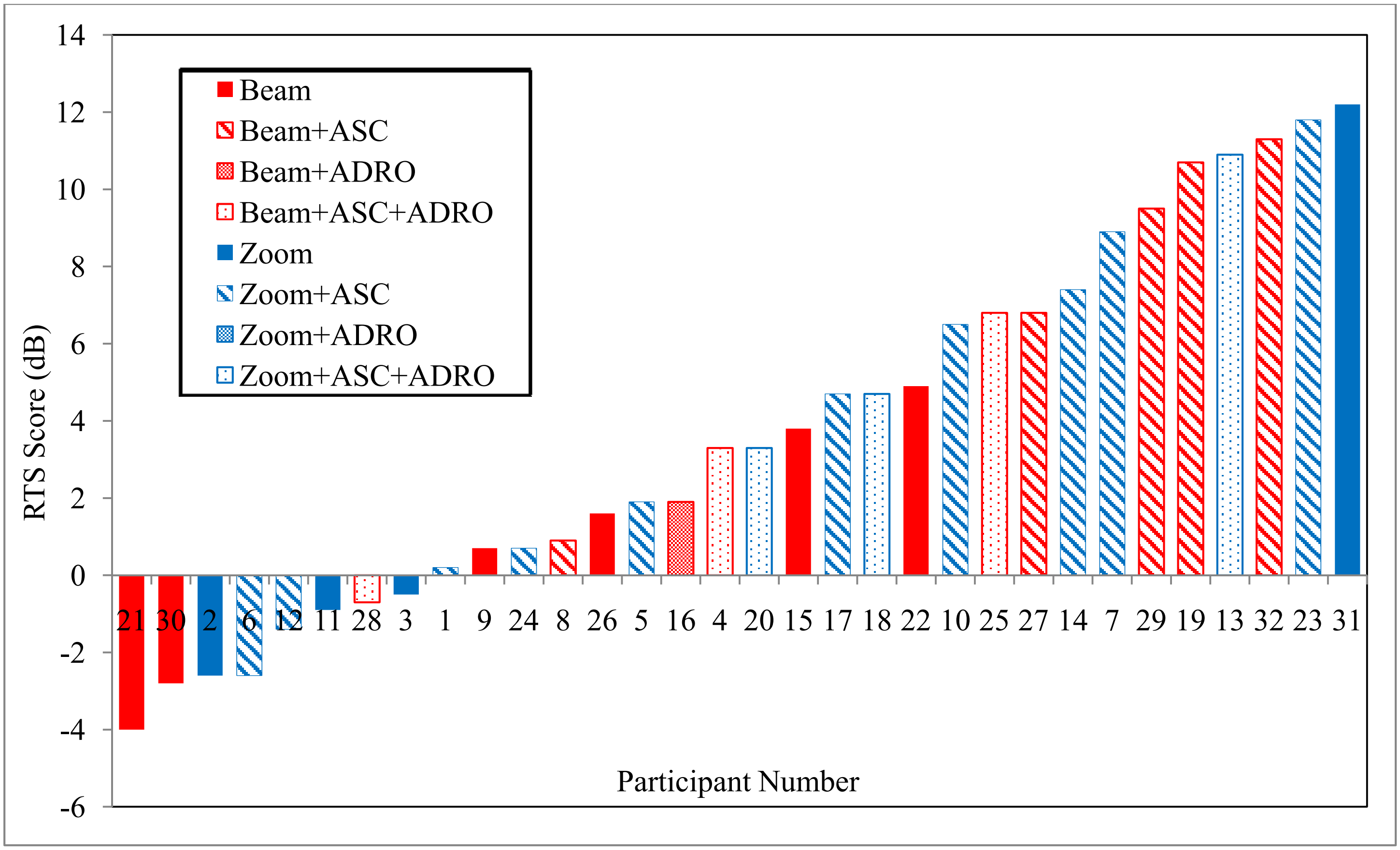

Table 4: Association among performance across processing options. Correlation coefficients (r) were computed for processing options within the Beam and Zoom groups separately.

Pearson Correlation Coefficients, N=32 Prob < |r| under H0: Rho=0			
	Beam+ASC	Beam+ADRO	Beam+ASC+ADRO
Beam	r = 0.79 p<.0001	r = 0.87 p<.0001	r = 0.79 p<.0001
Beam+ASC		r = 0.88 p<.0001	r = 0.84 p<.0001
Beam+ADRO			r = 0.82 p<.0001
	Zoom+ASC	Zoom+ADRO	Zoom+ASC+ADRO
Zoom	r = 0.77 p<.0001	r = 0.81 p<.0001	r = 0.75 p<.0001
Zoom+ASC		r = 0.73 p<.0001	r = 0.72 p<.0001
Zoom+ADRO			r = 0.88 p<.0001