Title of the Project: Pre-processing strategies and Speech perception in Cochlear implant users

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Abstract

The aim of the study was to investigate whether pre-processing strategies are beneficial for the perception of speech in the presence of noise in children using cochlear implants. Further, the study also aimed to determine whether there exists any difference in speech perception with the use of different pre-processing strategies such as Adaptive Dynamic Range Optimization (ADRO), Autosensitivity Control (ASC) and two stage adaptive beam forming algorithm (BEAM) in different signal to noise ratios (SNRs). Eighteen children aged 5 to 13 years, using Nucleus cochlear implants for at least 1 year served as participants. Their speech identification scores were tested in quiet with 'Everyday' default setting activated. They were also tested using speech noise at +5 dB and +10 dB SNR with ADRO, Beam, and ASC activated. A significant difference was found between the performance in quiet (with the 'Everyday' default setting) and in the presence of noise (with ADRO, ASC & BEAM). No significant difference was found between the 3 pre-processing strategies at both SNRs. The scores also did not differ between the 2 SNRs for all 3 pre-processing strategies. Thus, it was concluded that in conditions where the signal and noise emerge from the front of the listener, no influence of the pre-processing strategies was seen. With small variations in SNR (+5 dB, & +10 dB), the pre-processing strategies perform similarly.

Key words: ADRO, ASC, BEAM, SNR

Introduction

Cochlear implant (CI) technology has been found to afford increasingly high levels of speech understanding. Research in the past two decades has shown considerable improvement in speech recognition in quiet in individuals using cochlear implants (Tyler & Moore, 1992; Skinner et al., 1994; Skinner, Holden, Holden, Demorest, & Fourakis, 1997; Rubinstein, Parkinson, Lowder, Gantz, Nadol, & Tyler, 1998; Fetterman, & Domico, 2002; Firszt et al., 2004; Spahr & Dorman, 2004). However, speech perception performance is reported to result in deterioration with increasing levels of background noise (Fetterman, & Domico, 2002; Schafer, & Thibodeau, 2004). Difficulty in understanding in the presence of background noise has been observed to be a cause for dissatisfaction among CI recipients, despite advancement of CI technology (Brockmeyer, & Potts, 2011). Therefore, improving speech understanding in challenging environments has remained one of the most important design objectives for new commercial CI systems.

The reason for poor performance in the presence of noise is partially attributed to the loss of spectral resolution. Fu, Shannon, and Wang (1998) found that an increase in the effective number of spectral channels improved speech reception in noise of cochlear implant users. However, Frieson, Shannon, Baskent, and Wang (2001) observed that due to channel interaction, there could be a practical limit to the effective number of independent stimulation channels. Hence, they opined that increasing the number of electrodes may not aid in better speech perception in noise.

Historically, CI manufacturers have focused on developing and refining sound coding strategies to improve performance in adverse listening conditions. The performance in noise has been studied for different speech processing strategies employed in different implant systems (Hochberg, Boothroyd, Weiss, & Hellman, 1992; Skinner, Holden, & Holden, 1995;

Kiefer et al., 1996; Parkinson, Parkinson, Tyler, Lowder, & Gantz, 1998). Several studies have indicated a marked improvement in speech performance in the presence of noise with the use of noise reduction algorithms in cochlear implant users (Van Hoesel, & Clark, 1995; James, Blamey, Martin, Swanson, Just, & Macfarlane, 2002; Gifford, & Revitt, 2010; Brockmeyer, & Potts, 2011). The currently available cochlear implant speech processors are reported to be equipped with preprocessing strategies (Brockmeyer, & Potts, 2011) and/or external accessories (Schafer, & Thibodeau, 2004) that are designed to enhance perception in noisy conditions. The strategies used in the implants to enhance speech perception in the presence of noise vary depending on the manufacturer and the model of the device. The use of preprocessing strategies has primarily been used by Cochlear limited. On the other hand, CIs manufactured by Advanced Bionics Corporation and Med El utilize speech processing strategies to augment perception in the presence of noise (Vermeire, Punte, & Van de Heyning, 2010; Kam, Yee, Cheng, Wong, & Tong, 2012).

In addition to the preprocessing strategies, the use of dual-port microphones has been integrated into the speech processor manufactured by Cochlear limited for many years (Patrick, Busby, & Gibson, 2006). As reported by Dillon (2001), in a dual-port directional microphone arrangement, sounds from behind reach the rear port before the front port, creating an external time delay. The external time delay is noted to depend on the distance between the two micro phone ports, which is 7 mm in the Nucleus Freedom device. The rear port is reported to have 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. It was noted that if the internal and external time delays were equal, sound from the rear would reach both sides of the microphone diaphragm at the same time, generating no net force and suppressing sounds from the rear direction. Also, Thompson (2002) observed that the direction of maximum suppression varied with the difference between the internal and external time delays.

A variety of noise reduction algorithms to improve speech perception in noise have been made available in the Nucleus Freedom processor and subsequent models of processors manufactured by Cochlear limited. The noise reduction algorithms used by Cochlear Limited that are incorporated in the pre-processing strategies have been termed as SmartSounds. This term represents the four input processing technologies available on the Nucleus Freedom speech processor [Adaptive Dynamic Range Optimization (ADRO), Autosensitivity control (ASC), a two-stage adaptive beam forming algorithm (BEAM) & Whisper] and the five in CP810 processor (ADRO, ASC, BEAM, Whisper, & Zoom). Each of the different SmartSound options is designed to help pre-process incoming sound in different ways for optimum benefit in different listening environments.

Adaptive Dynamic Range Optimization (ADRO) was reported by Blamey, James, Wildi, McDermott, and Martin (1999) to be an adaptive system that adjusts the channel gains to ensure speech is always delivered at a comfortable listening level, for changing listening situations. James et al. (2002) described ADRO as a process that adjusts the gain of the input signal across the frequency range so that the product of the signal processing matches specified targets in the upper part of a 30 dB input range. The processing was described to be multi-channel, as the gain adjusts in individual frequency filters. It was implemented digitally so that the gain increased or decreased through compliance with a number of conditional rules. The gain increased linearly if a sound was within the audible and comfortable range. ADRO adjusted the input signal so that the output was comfortably loud. According to James et al. (2002) and Dawson, Decker, and Psarros (2004) the gain did not exceed a specified maximum amount. This maximum gain rule worked to limit the amplification of low-level background noise. Patrick, Busby and Gibson (2006) reported that

ADRO was incorporated into the Nucleus CI system in 2002 as an input signal processing option. However, in the CP810 processor, according to a company report, ADRO has been incorporated after the filtering process, permitting independent adjustments of each filter band. This is unlike their front-end processing (ASC, BEAM, Whisper, & Zoom), that is reported to act on all the electrodes prior to the filtering process.

James et al. (2002) investigated the acceptability and the effect of ADRO on speech perception in nine individuals with cochlear implants. Speech processor programs were created with and without ADRO. Speech perception performance was compared for the standard and ADRO programs using City University of New York (CUNY) sentences, consonant-nucleus-consonant (CNC) words, and closed set spondees presented in quiet using a range of presentation levels from 70 to 40 dB SPL. The testing was carried out in multi-talker babble with 15 dB and 10 dB SNRs using CUNY sentences. The results revealed a significant improvement in speech perception scores with the ADRO programs over the standard. At 50 dB, the mean open set sentence scores in quiet improved by 16% ($p < 0.001$); at 60 dB the mean CNC word score improved by 9.5% ($p < 0.001$); and at 40 dB the mean spondee score increased by 20% ($p < 0.05$). However, no significant difference was noticed between the ADRO and the standard program for sentences presented in either noise conditions. The authors opined that continual adjustment of channel gains using ADRO provided improved sound quality and improved speech perception performance. Hence, they reported that ADRO could be a viable alternative to fixed channel gain.

Autosensitivity Control (ASC) was described by Brockmeyer, and Potts (2011) to be an optional processing scheme that automatically adjusts the sensitivity according to the noise floor, or the intensity level of sound during breaks in speech. They reported that the ASC processing strategy was mainly developed to reduce the usage of the manual sensitivity control by CI recipients. This input preprocessing strategy was observed to provide substantial benefit in speech perception in noise. The effect of ASC on speech performance in quiet and noisy situations was evaluated in ten individuals using Nucleus Freedom implant by Wolfe, Schafer, Heldner, Mulder, Ward, and Vincent (2009). Sentences were presented from a loudspeaker which was placed at an angle of 0^0 and noise originated from loudspeakers placed at the 4 corners of the room. The level of stimulus presentation was 60 dB A in quiet, 65 dB A with a +10 dB SNR, 70 dB A with a +7 dB SNR, and 75 dB A with a +4 dB SNR. The results showed that the participants performed significantly better in the +7 and +4 dB SNR conditions with ASC on. Speech recognition with ASC with noise at 65 and 70 dB A was significantly better than in all conditions with no ASC as well as with ASC with noise at 75 dB A.

Beam was introduced in the Nucleus Freedom speech processor in 2005, as reported by Spriet et al. (2007). According to them, beam combined information from both a front directional and a rear omni directional microphone on the processor. The directional microphone system was reported to contain two ports separated by 0.7 cm. The rear omni directional microphone is separated from the front port of the directional microphone by 1.9 cm. According to Patrick, Busby, and Gibson (2006), Beam works in two phases, the first being a directional operation and the second phase effects an adaptive noise cancellation operation. Beam was recommended for specific listening situations in noise where the sound source is in the front of an individual and the interfering noise sources are at the sides and/or behind the individual.

As early as 1995, Van Hoesel, and Clark used a bilateral two-microphone adaptive beamformer (ABF) to evaluate the effect of noise reduction algorithms in cochlear implants. They demonstrated a directional gain of about 10 dB with Nucleus 22 cochlear implantees. Tests were conducted in a sound proof booth with the target speech presented in the front of the patient and multi-talker babble noise presented at 90^0 to the left. These acoustical test conditions were considered to simulate real life living room environments. The CVC identification and SRT for numbers were evaluated. The results revealed that the two-microphone and two-stage adaptive filtering strategy lead to very significant improvements in CVC identification and SRT in the presence of steady and non-steady noise. With the ABF a score of 80% was obtained in quiet and 40% in noise. A significant improvement of 30% was obtained with ABF in noise. The authors concluded that for speech-in-noise applications, a directional microphone would perform better than an omnidirectional microphone.

Spriet et al. (2007) evaluated the performance of 5 CI users using Nucleus freedom speech processor. Performance of the Beam processing strategy was assessed using repeated sentences 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 either from a single source situated at an angle of 90^0 or from three sources located at 90^0, 180^0, and 270^0 azimuths. The findings of the study indicated that Beam improved the average SNR in all conditions when compared to the standard directional microphone.

Whisper is yet another noise reduction algorithm that has been implemented in the SmartSound strategy. It was described to be an additional compression circuit at the input stage of the processor, to enhance soft and distant sounds as reported in Nucleus Freedom clinical outcomes (2006). The benefits of Whisper in the Nucleus Freedom speech processor were investigated in a study by the Cooperative Research Centre for Cochlear Implant and Hearing Aid Innovation (Nucleus Freedom Clinical Outcomes, 2006). Eight adult cochlear implantees who used the Freedom speech processor participated in the study and were tested under three different conditions using one program with Whisper and another program without any pre-processing options. Tests included CNC words presented at 50 dB SPL and

60 dB SPL and CUNY sentences in noise at 70 dB SPL (+10 dB SNR). A statistically significant improvement of 11% was obtained with Whisper for CNC words presented at 50 dB SPL in quiet. There was no significant difference in speech performance scores with Whisper when CNC words were presented at 60 dB SPL and Whisper was not favourable when used for CUNY sentences in noise.

Zoom is a noise reduction algorithm which is found to be available with the current generation sound processor, CP810 of Cochlear Limited. According to the Clinical results with the New Cochlear Nucleus 5 system (2010), Zoom uses a dual microphone system which enhances speech performance in noise. The directionality pattern, unlike Focus that uses the BEAM algorithm, is reported to not vary as a function of changes in spatial separation between speech and noise, keeping the soundscape relatively constant. The Zoom directional pattern is designed to give optimal blocking of diffuse noise, or when noise comes from many directions simultaneously.

Studies have been carried out using a combination of preprocessing strategies to check their utility in noise reduction (Wolfe, 2010; Gifford, & Revitt, 2010). In addition, these studies have also compared noise reduction strategies across different CI companies. Wolfe (2010) compared the performance of Nucleus freedom cochlear implant recipients and Advanced Bionics Clarion II cochlear implant recipients in typical listening conditions. The adult participants included eight unilateral Nucleus users, four bilateral Nucleus users, six unilateral Advanced Bionics users and six bilateral Advanced Bionics users. Speech recognition in quiet was assessed at 60 dB SPL with two full lists of the CNC monosyllabic word recognition test. Additionally, speech recognition in noise was assessed with four full lists of the BKB-SIN test to provide an average SNR for 50% correct performance. The target sentences of the BKB-SIN test were presented at 75 dB SPL, with an initial SNR of +21 dB for the first sentence of each list which decreased by 3 dB for each successive sentence in the list. The participants using Nucleus were tested with ASC+ADRO and the Advance Bionic participants used an input dynamic range (IDR) varying between 60 and 65. The t-test revealed a significantly better performance ($p = 0.01$) for the combined group of Nucleus recipients using ASC+ADRO compared to the combined group of Advanced Bionics recipients for the BKB-SIN sentence test in noise. Significant better performance ($p = 0.02$) was also observed for the unilateral Nucleus recipients using ASC+ADRO (89%) compared to unilateral Advanced Bionics recipients (79%) for the same test. A trend towards better performance was observed for the bilateral Nucleus recipients using ASC+ADRO compared to the bilateral Advanced Bionics recipients for the BKB-SIN sentence in noise. Similarly, a trend towards better performance was also observed for unilateral Nucleus recipients using ASC+ADRO (85%) compared to unilateral Advanced Bionics recipients (76%) for the CNC word test in quiet. The author reported that the input processing scheme of the Nucleus cochlear implant system consisting of the ASC+ADRO algorithms provided better performance in everyday listening conditions. The study highlighted the importance of using input processing schemes, in particular the ASC+ADRO algorithms combined, for good speech perception by CI recipients in everyday listening conditions.

Gifford, and Revitt (2010) assessed speech perception for adult cochlear implant users in the presence of a realistic restaurant simulation generated by an eight-loudspeaker R-SPACE array in order to determine whether commercially available preprocessing strategies and/or external accessories yielded improved sentence recognition in noise. Thirty-four subjects, ranging in age from 18 to 90 years, participated in the study. Fourteen subjects were Advanced Bionics recipients, and 20 were Cochlear Corporation recipients. SRTs in noise were assessed with the participants' preferred listening programs as well as with the addition of either BEAM of Cochlear Corporation or the T-Mic accessory option of Advanced Bionics. Adaptive SRTs with the Hearing-in-Noise-Test sentences were obtained for all 34 subjects. In addition, 16 of the 20 Cochlear Corporation subjects were reassessed obtaining an SRT in noise using the combination of noise reduction algorithms: ADRO (Everyday SmartSound environment), ADRO+ASC (Noise SmartSound environment) and ADRO+ASC+BEAM (Focus SmartSound environment). Statistical analysis revealed that in the Cochlear Corporation recipients, both Noise and Focus SmartSound environment resulted in significant improvement in SRT in noise when compared to the Everyday SmartSound environment. The Focus SmartSound programme yielded equivalent or better performance in noise compared to the Noise SmartSound programme. The degree of improvement in the SNR ranged from 0 to 7.33 dB. The mean SRT performance for the preferred program and Focus was 11.2 and 7.3 dB SNR, respectively, revealing that the mean improvement in the SNR for all 20 subjects was 3.9 dB. Further, it was also observed that the T-Mic accessory option in Advanced Bionics significantly improved the SRT when compared to the BTE mic. For all 14 subjects, the T-Mic yielded equivalent or better performance in noise. The degree of improvement in the SNR with the use of the T-Mic ranged from 1.3 to 8.3 dB. It was concluded that the Focus SmartSound environment and the T-Mic resulted in similar degrees of improvement that were not found to be significantly different from one another.

Laboratory measures of noise reduction algorithms are often found to show greater levels of improvement than reported by participants in everyday listening situations. Hence, to address this issue, Brockmeyer, and Potts (2011) measured speech recognition of 27 unilateral and three bilateral adult Nucleus Freedom CI recipients in R-SPACE with four processing options: standard dual-port directional (STD), ADRO, ASC, and BEAM at two noise levels. The R-SPACE test system, developed by Compton-Conley and colleagues, was considered to replicate a restaurant environment. The R-SPACE consisted of eight loudspeakers positioned in a 360^0 arc through which a recording of a restaurant background noise was played. The participants' everyday program (with no additional processing) was used as the STD program while ADRO, ASC, and BEAM were added individually to the STD program to create a total of four programs. Hearing-in-Noise-Test sentences were presented at 0^0 azimuth with R-SPACE restaurant noise at 60 and 70 dB SPL. The reception threshold for sentences (RTS) was obtained for each processing condition and noise level. The results showed that in 60 dB SPL noise, STD processing resulted in a mean RTS of 10.8 dB. The poorest performance was with ADRO, with a mean RTS of 12.8 dB. ASC and BEAM processing showed an improvement in RTS relative to STD and ADRO processing, with means of 9.5 and 8.3 dB, respectively. In 70 dB SPL noise, ASC (10.2) and BEAM (12.2) had significantly better mean RTSs compared to STD (15.6) and ADRO (15). Comparison of noise levels showed that STD and BEAM processing resulted in significantly poorer RTSs in 70 dB SPL noise compared to the performance with these processing conditions in 60 dB SPL noise. Bilateral participants demonstrated a bilateral improvement compared to the better monaural condition for both noise levels and all processing conditions, except ASC in 60 dB SPL noise. The authors suggested that the use of processing options involving noise reduction would improve a CI recipient's ability to understand speech in noisy environment.

The effect of front-end processing on cochlear implant performance of children was evaluated by Wolfe, Schafer, John and Hudson (2011). The authors investigated the potential benefits of ADRO as compared with ASC+ADRO for children using Nucleus 5/Freedom cochlear implants. Eleven subjects aged 4 years to 12 years with unilateral or bilateral cochlear implants participated in the study. Speech perception of PBK-50 monosyllabic words in quiet and BKB-SIN sentences in noise was measured for each participant. The data in quiet was analysed using descriptive statistics and the conditions in noise were compared using 1-way repeated measures. In the quiet situation the participants obtained scores at or above 90%. In noise, sentence perception performance in the ASC+ADRO condition was significantly better than that in ADRO alone condition. The average speech-in-noise threshold was 8.9 and 5.5 for ADRO and ADRO+ASC respectively. The authors suggested considerable benefit from a combination than the sole strategy implementation in speech in noise performance.

In Med-El cochlear implant users, Vermeire, Punte, and Van de Heyning (2010) investigated the long-term effects of a new Fine Structure Processing (FSP) speech coding strategy over High Definition Continuous Interleaved Sampling (HDCIS) on speech perception in noise and on the quality of life. Both these strategies were assumed to provide fine spectral information by using filters with bell-shaped frequency response. The FSP strategy, additionally provided temporal fine structure by using stimulations at the 1 – 3 most apical electrodes that are elicited at a variable rate that corresponds to the fine structure of the signal in the specific filter band. According to Hochmair et al, 2006, the purpose of the FSP strategy was to try to provide CI users with improved pitch perception, which in theory would improve speech recognition (especially in noise), music appreciation, and sound localization. Thirty-two users of the Med-El Pulsar CI 100 system, who had switched over from TEMPO+ to OPUS 2 speech processor, took part in the study. Of them, 22 used FSP and the remaining used high-definition continuous interleaved sampling (HDCIS). The subjects were tested with the Tempo+ with CIS+ strategy just before the switch-over and after 12 months of use with OPUS 2 having FSP/HDCIS strategy. Performance with FSP/HDCIS was tested at switch-over, and after 1, 3, 6 and 12 months. Tests used were a sentence-in-noise test, Speech Spatial and Qualities of Hearing Scale (SSQ) questionnaire. The results showed that in the FSP group, the SRT showed a deterioration of 3.3 dB at the acute switch-over interval (< 1 month), but demonstrated a significant improvement over time ($p < 0.001$) with a final benefit of 6.5 dB after 12 months of FSP use. A significant improvement over time can also be seen on the spatial sub-score of the SSQ questionnaire ($p = 0.009$). No significant differences were seen in the HDCIS group. The experimenters inferred that by enhancing fine structure coding in the lower frequencies, as implemented in the FSP coding strategy, speech perception in noise could be enhanced.

The impact of different speech coding strategies available in Med-El cochlear implants on speech perception in quiet and in the presence of noise was studied by Magnusson (2011). Fine Structure Processing (FSP) and High Definition CIS (HDCIS) were compared in 20 experienced adults. These participants were upgraded to FSP from CIS+. Blinded paired-comparisons between FSP and HDCIS were performed for speech intelligibility and music sound quality. The results showed no significant differences between the strategies. However, for speech stimuli, 11 of the participants preferred FSP and 9 preferred HDCIS while for music, 5 preferred FSP and 15 preferred HDCIS. In addition, the average speech recognition scores decreased significantly after one month with FSP, but after two years, no significant difference was observed compared to the initial results obtained with CIS+. The author concluded that the recipients should be given the choice of choosing between the strategies owing to the large individual differences in subjective preference, and the fact that the FSP strategy was not superior to the CIS variations.

Advanced Bionics Corporation developed a new strategy ClearVoice, which is reported to be a software algorithm that could be implemented with HiRes 120 on the Harmony behind-the-ear (BTE) processor. It is found to identify frequency bands in which unwanted sound is present, and thereby reduce transmission of information in those bands enhancing the desired signal perception (Kam et al., 2012; Litvak, Spahr, Saoji, & Fridman, 2007). Kam et al. (2012) reported that the strategy is designed to improve listening in adverse listening environments without compromising performance in quiet environments.

A pilot study was carried out by Buechner, Brendel, Saalfeld, Litvak, Frohne-Buechner, and Lenarz (2010) to evaluate the efficacy of signal enhancement algorithms on sentence perception in the presence of noise. The participants, who were 13 post-lingually deafened adults using HiRes 120 cochlear implants, were fitted with 2 versions of the noise reduction algorithm, ClearVoice; One version had a moderate setting that had -12 dB reduction of noise and the other had a strong setting that resulted in -18 dB reduction. Hochmair-Schulz-Moser (HSM) sentence test in speech-shaped noise was administered using the clinical program as well as both noise reduction programs. It was seen that during the fitting, all participants exhibited no difficulties and accepted the noise reduction programs without any acclimatization. They achieved significantly better results with both noise reduction programs on the HSM sentence test in noise compared with the clinical program. Group mean speech perception scores were highly significantly better for the ClearVoice settings compared with the clinical program score. The study demonstrated the benefit of the noise reduction algorithms in Advanced Bionic cochlear implants processors.

Kam et al. (2012) evaluated the benefits of ClearVoice strategy on speech perception in noise and in everyday listening situations in twelve adult Cantonese speaking cochlear implant users. Participants used harmony implant and HiRes 120 sound processing strategy. The procedure included carrying the Cantonese Hearing-in-Noise-test in two conditions, with and without ClearVoice strategy. They were also evaluated after a week of acclimatization with the medium and high ClearVoice gain settings. The results revealed that the performance of the participants improved with ClearVoice turned on over it turned off in the presence of noise ($p<0.05$). However, there was no significant difference in performance across the three different settings, without the ClearVoice, ClearVoice medium and ClearVoice high in quiet.

Another option available with Advanced Bionic is the T-Mic. As reported in the Advanced Bionics website [Advancedbionics/Products/T-Mic (n.d.)], it is found to aid in natural directional hearing, listening in difficult situations, natural use of a telephone, natural use of headphones and protection from wind noise. The T-Mic as reported is designed to enable the individual to hear better in common noisy environments, such as classrooms, restaurants, and sporting events. Gifford, and Revitt (2010) noted that the T-Mic accessory option in Advanced Bionics significantly improved SRT when compared to the BTE mic in the 14 participants studied by them. The T-Mic was found to result in equivalent or better performance in noise.

The literature on noise reduction algorithms in individuals with cochlear implants has indicated benefit in speech understanding in adverse listening conditions. However, the researchers have mainly focused on studying the impact of isolated strategies (James et al., 2002; Spriet et al., 2007) or comparing combinations of pre-processing such as ADRO+ASC or ASC+BEAM (Gifford, & Revitt, 2010; Wolfe, 2010). This makes it difficult to compare the impact one strategy over the other. Therefore, it is necessary to evaluate the effectiveness of individual pre-processing strategies on speech perception in CI users. There is also a need to determine how the different strategies function in the presence of different SNRs. This would shed light on whether the strategies function in a similar or different manner with varying SNRs. In addition, the majority of studies reported in literature have utilized small samples of CI users based on which they draw their conclusion. There is an urgent need to check the utility of different preprocessing strategies on a larger population.

Further, most of the reported literature on the utility of noise reduction algorithms has been evaluated on post-lingual adults (Spriet et al., 2007; Brockmeyer, & Potts, 2011). Such data may not necessarily reflect the utility on children. Post-lingual adults would have the capacity to utilize redundant cues in order to perceive acoustical signals that they may miss out in adverse listening conditions. However, children who do not have past linguistic exposure like adults are more likely to not use these redundant cues. Thus, there is a need to study the impact of noise reduction algorithms on children.

Hence, the main aim of the study was to investigate whether noise reduction algorithms were beneficial for the perception of speech in the presence of noise in children using cochlear implants. Further, the study also aimed to determine whether there exists any difference in speech perception with the use of different noise reduction algorithms such as ADRO, ASC and BEAM in different SNRs. Such information would serve as a basis for recommending the use of specific noise reduction algorithms in specific listening conditions.

Method

Participants

Eighteen individuals using Nucleus cochlear implants for at least 1 year with stable maps participated in the study. All the participants, aged 5 to 13 (mean age of 8; 7 years), had congenital hearing impairment. They had been exposed to Kannada (N = 16) or Indian English (N = 2) from early childhood, the former being a language spoken in South India. They used Nucleus 22/24/512/Freedom implants with SPrint (N = 5), ESPrit 3G (N = 1), Freedom (N = 6) or CP810 (N = 6) sound processors that had facilities to activate various pre-processing strategies. ACE was the speech coding strategy used by all the participants. Demographic details of the participants are provided in Table 1. All the participants had aided thresholds within the speech spectrum and all but one had speech identification scores greater than 50% in quiet. Only one of the participants had an aided speech identification score of 44% in quiet, though the aided thresholds were well within the speech spectrum. The open-set speech identification scores in quiet of the participants, with their regularly used settings, ranged from 44% (11/25) to 88% (22/25) with the mean being 66.44% (16.61/25). It was ensured that the participants had no other neurological or otological symptoms and were able to provide consistent responses. The participants had a minimum of 6 months experience with hearing aids prior to the use of implants. Only one participant had no exposure to a hearing aid before undergoing implantation (Table 1).

Prior to the commencement of the study, informed consent was taken from the caregivers of the participants. It was also ensured that the recommendations of the 'Ethical Guidelines for Bio-Behavioural Research Involving Human Subjects' (2003) of the All India Institute of Speech and Hearing were adhered to.

Table 1: *Demographic details of the participants*

Client no.	Age in Years	Gender	Implant	Speech processor	Experience with CI (in years)	Years of initial hearing aid usage	Open set SIS in quiet
1.	5	F	Freedom CA	CP810	2	1	80% (20/25)
2.	8	M	CI512	CP810	1	3	60% (15/25)
3.	5	F	CI512	CP810	2	1;5	80% (20/25)
4.	6	M	CI512	CP810	1;4	2	80% (20/25)
5.	13	F	Nucleus 24	CP810	7	1	88% (22/25)
6.	6	F	CI512	CP810	1	2	56% (14/25)
7.	9	F	Nucleus 24	Freedom	1	1	80% (20/25)
8.	12	F	Freedom CA	Freedom	4	4	72% (18/25)
9.	10	F	Freedom CA	Freedom	4	1	44% (11/25)
10.	6	F	Freedom CA	Freedom	2	1	72% (18/25)
11.	9	M	Nucleus 24	Freedom	4	2;6	68% (17/25)
12.	7	M	Freedom CA	Freedom	3	2;6	56% (14/25)
13.	12	F	Nucleus 24	SPrint	3	1;6	68% (17/25)
14.	12	F	Nucleus 24	SPrint	6	1;6	76% (19/25)
15.	10	F	Nucleus 24	SPrint	4;5	<1	64% (16/25)
16.	9	M	Nucleus 24	SPrint	4	1	72% (18/25)
17.	13	F	Nucleus 24	SPrint	4	3	64% (16/25)
18.	6	F	Nucleus 22	ESprit 3G	3	0	84% (21/25)

Test equipment and material

Custom Sound version 3.2 developed by Cochlear Limited was used to program the speech processor of the participants. The programming was carried out by two audiologists who had over 8 years of experience doing cochlear implant mapping. The speech processor of each participant was hardwired through a programming interface (Portable Programming System/ Programming Pod, depending on the type of the processor) to a personal computer loaded with the Custom Sound software. The same software was used to implement preprocessing strategies in the speech processor of all the participants along with their standard map.

A calibrated double channel diagnostic audiometer, Orbiter 922 (version 2) was used to carry out the speech perception tests. A loud speaker, calibrated to present noise and speech at different SNRs was positioned at a distance of 1 meter from the participant at an angle of 0^{o} azimuth.

The participants who spoke Kannada (N = 16) were evaluated on the ‘Kannada phonemically balanced word identification test’ (Yathiraj, & Vijayalakshmi, 2005) and those who spoke English (N = 2) were evaluated using the ‘Monosyllable speech identification test in English for Indian children’ developed by Rout (1996). The Kannada test has 4 lists of 25 words each which are familiar to children aged 5 years and above. The material within each list was randomized to avoid word familiarity effects which led to the formation of 8 lists. The Rout test consists of 2 lists each having 25 phonemically balanced words with norms established on children. These words were randomized to form additional lists. A personal computer, connected to the auxiliary input of the audiometer, was used to present the recorded speech material.

Test environment

The testing procedure was carried out in an air-conditioned sound treated suite. The permissible noise limits as in the test facility was as per ANSI standards (S3.1-1991).

Procedure

Aided warble tone thresholds were obtained at octaves and mid octaves (250 Hz to 8 kHz) using a modified Hughson-Westlake procedure. The measurement was carried out with the participants seated at 1 meter from the loudspeaker which was placed at 0^{o} azimuth. Further testing was done only if the aided thresholds were found to be well within the speech spectrum.

The aided speech identification performance of the participants was tested in 4 conditions. The conditions included the ‘Everyday’ default setting and 3 pre-processing strategies (ADRO, ASC, & BEAM). The ‘Everyday’ default setting was tested in quiet and the 3 pre-processing strategies were tested in 2 signal-to-noise ratios (+5 dB & +10 dB SNR) using speech noise. A participants’ ‘Everyday’ default setting varied depending on the type of the speech processor they used. For the CP810 processor, the default setting was ADRO+ASC, whereas for the Freedom, SPrint and ESPrit 3G processors, the default setting was ADRO. The speech identification abilities of the participants were tested with the speech processor activated with one algorithm at a time. The recorded speech tests as well as the speech noise was presented through the same loudspeaker at 0^{0} azimuth.

Speech identification testing was done with the recorded speech test material presented at 45 dB HL that corresponded to a normal conversational level. The stimuli were played using a personal computer connected to the auxiliary input of the audiometer. Speech noise was generated from the audiometer. All the participants were initially tested in the quiet situation followed by the +5 dB SNR and +10 dB SNR noise conditions. The order in which they were tested with the pre-processing strategies (ADRO, ASC, & BEAM) was randomized to avoid any test order effect. The participants were instructed to listen to the speech stimuli and give an oral response. Written responses were obtained if a child had misarticulation. All testing was done within one session. Breaks were given if a child was found to be restless. Appropriate reinforcements were provided to the participants. Test-retest reliability was determined by repeating the procedure on two participants after an interval of two months.

The obtained scores were tabulated and analyzed to determine the performance of individuals using cochlear implants in quiet and in presence of noise across the three pre-processing strategies (ADRO, ASC and BEAM). The data was subjected to statistical analyses using SPSS software (Version 18).

Results

The speech identification scores obtained by the eighteen participants were analysed using SPSS software (Version 18). The analysis was done for their responses in quiet with them using their 'Everyday' default setting. Additionally, their responses with the activation of 3 pre-processing strategies (ADRO, ASC, & BEAM) in 2 noise conditions (+5 dB, & +10 dB SNR) were analysed. Initially, the data were analysed without and with the scores of the 2 children who were tested in English. This was done to check if the language of evaluation made a difference in the statistical output so as to decide whether the scores of the 2 children should be included or excluded from further analyses. Inclusion of the 2 English speaking children was necessary to statistically determine the influence certain parameters such as the effect of microphone directionality on the perception of speech.

Table 2: *Mean, SD and Confidence intervals for 16 participants*

Strategy	SNR	Mean*	SD	Confidence interval		N
				Lower bound	**Upper bound**	
Everyday (default)	Quiet	17.31	3.15	15.58	19.54	16
ADRO	+ 5 dB	13.56	3.86	11.83	15.79	16
	+ 10 dB	12.75	4.62	11.08	15.04	16
ASC	+ 5 dB	13.25	4.50	11.58	15.54	16
	+ 10 dB	13.43	4.27	11.65	15.60	16
BEAM	+ 5 dB	13.10	3.69	10.80	15.80	10
	+ 10 dB	14.40	4.62	11.80	16.80	10

Note. * *Maximum score = 25*

Table 2 shows the mean and the standard deviation (SD) of the speech identification scores of the 16 Kannada speaking participants in quiet and noise, at two SNRs across the 3 processing strategies. Of the 16 participants only 10 had provision for BEAM in their processor. Likewise, Table 3 depicts the score of all 18 participants, which includes the 16 Kannada speaking children and the 2 English speaking children. Among the 18 participants, BEAM could be activated in 12 of them. From Table 2 and 3, it is clear that the mean and SD did not vary much without and with the addition of the two 2 English speaking children for all strategies and SNRs.

Table 3: *Mean, SD and Confidence intervals for 18 participants*

Strategy	**SNR**	**Mean***	**SD**	**Confidence interval**		**N**
				Lower bound	**Upper Bound**	
Everyday (default)	Quiet	17.33	2.9	15.47	19.20	18
ADRO	+ 5 dB	13.77	3.68	11.91	15.64	18
	+ 10 dB	13.00	4.40	11.02	14.75	18
ASC	+ 5 dB	13.50	4.38	11.64	15.37	18
	+ 10 dB	13.55	4.06	11.69	15.42	18
BEAM	+ 5 dB	13.50	3.55	10.97	15.53	12
	+ 10 dB	14.47	4.25	12.05	16.63	12

Note. * *Maximum score = 25*

Further, Mauchly's test of sphericity was used to check if the data obtained from 16 children (excluding the 2 English speaking children) and the 18 participants (including the 2 English speaking children) met the assumptions required to carry out the analysis of variance. The results of the Mauchly's test showed that the sphericity was assumed in both the data set ($W = 1$, $p > 0.05$). Hence, the assumptions to carry out ANOVA were satisfied in both the data set.

Additionally, it was found that the mean scores of the participants without and with the addition of the 2 English speaking children had similar confidence intervals (Table 2 &

3). This was observed for all three noise conditions (quiet. +5 dB SNR, & + 10 dB SNR) and 3 pre-processing strategies (ADRO, ASC, & BEAM).

Figures 1 and 2 depict the individual raw speech identification scores of the 18 participants in the quiet condition with them using the ‘Everyday’ default setting and for the 3 noise reduction algorithms (ADRO, ASC, BEAM). Figure 1 provides information of performance at the +5 dB SNR and Figure 2 at the +10 dB SNR. Participants 1 to 6 were the CP810 users, 7 to 12 were Freedom users, 13 to 17 were SPrint users and 18 was the ESPrit 3G user.

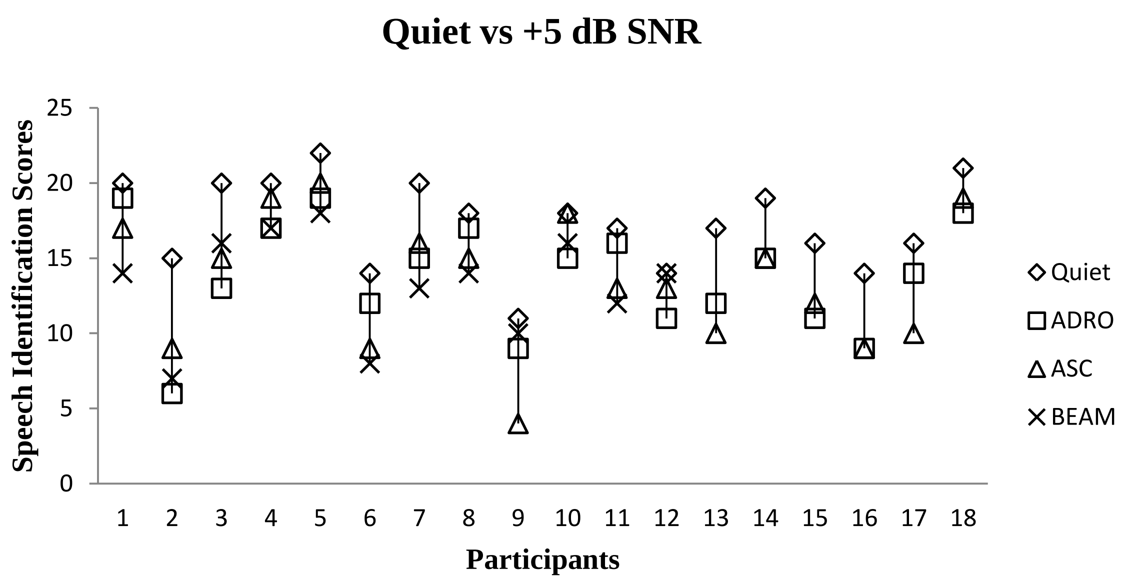

Figure 1: Speech identification scores of participants in quiet and at +5 dB SNR across pre-processing strategies

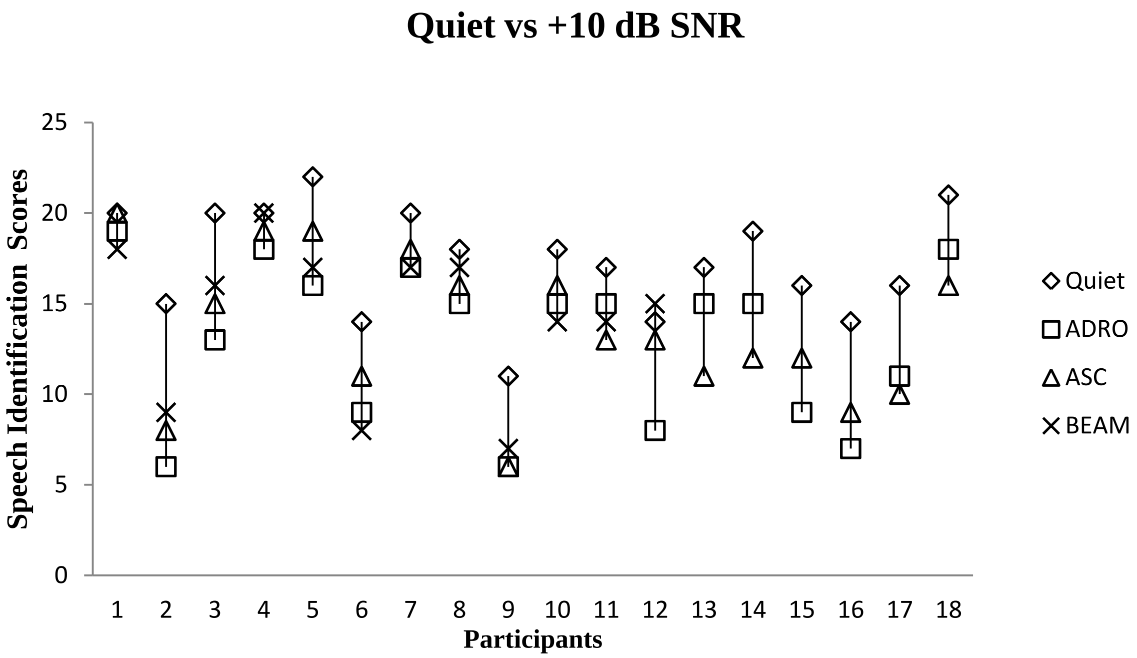

Figure 2: Speech identification scores of participants in quiet and at +10 dB SNR across pre-processing strategies

Thus, the comparison of the analysis without and with the 2 English speaking children revealed that there were only marginal differences in the mean and SD values, sphericity was met in both analyses and there was no significant difference between the mean scores. This indicated that the inclusion of the scores of the 2 English speaking children did not alter the results. Hence, all further analyses were carried out with the scores of all 18 participants grouped (16 Kannada speakers, & 2 English speakers). Thus, the data were analysed for the 18 participants to compare the speech identification scores without and with the presence of noise; compare the speech identification scores across the three pre-processing strategies; compare the effect of the 2 SNRs; and compare the speech identification scores as function of microphone directionality.

i. Comparison of scores between default setting in quiet and pre-processing strategies

To compare the performance of the participants in quiet with their performance in noise with different pre-processing algorithms activated, one-way repeated measure ANOVA and paired t-test were used. One-way repeated measures ANOVA was used when analysing scores obtained on 2 of the pre-processing strategies (ADRO & ASC) that had 18 participants. Due to the unequal number of participants using BEAM and the other two strategies, BEAM could not be analysed using ANOVA. Hence, paired t-test (2 tailed) was used when analysing data of BEAM that was obtained only from 12 participants.

One-way repeated measures ANOVA with 2 pre-processing strategies (ADRO & ASC) combined as well as the SNRs combined, a significant main effect was seen for the 18 participants [$F(1, 34) = 1.106$, $p < 0.05$]. One-way ANOVAs done for each of the SNRs also revealed a similar significant main effect for the +5 dB SNR condition [$F(3, 34) = 1.234$, $p < 0.05$] and the +10 dB SNR condition [$F(3, 34) = 4.95$, $p < 0.05$].

The output from the one-way ANOVA was analysed using Bonferroni multiple comparison test. This was done to check if there existed any significant difference between the scores got in the quiet condition and with ADRO and ASC activated. The paired-wise comparison revealed that there was a significant difference between the scores got in the quiet condition with the participants using their default settings and in the presence of noise (with SNRs combined) for ADRO ($p < 0.01$) and ASC ($p < 0.01$). Also, at each of the SNRs, the paired-wise comparison between the scores obtained in quiet and the 2 algorithms were statistically significant at the 0.01 level.

The paired t-test carried out to check the difference in scores obtained in quiet with that got with BEAM for 12 participants was statistically significant. This significant difference was observed for the +5 dB SNR condition [$t(11) = 6$, $p < 0.01$] and +10 dB SNR condition [$t(11) = 4.78$, $p < 0.01$].

ii. Comparison of scores across the pre-processing strategies

Two-way repeated measure ANOVA was carried out to check the impact of ADRO and ASC across the 2 SNRs (+5 dB & +10 dB). The scores obtained with the participants using BEAM could not be tested using ANOVA as this algorithm could be activated on only 12 of the 18 participants. The 2-way repeated measure ANOVA (2 algorithms x 2 SNRs) indicated that there was no significant main effect [$F(1, 34) = 0.074$, $p > 0.05$] when the two SNRs were combined. Since no significant main effect was seen and no interaction between the 2 pre-processing strategies and the 2 SNRs, no further analysis was carried out.

To compare the scores obtained using BEAM with ADRO and ASC, independent t-test was carried out. The results of the t-test revealed that there was no significant difference in performance between ADRO and BEAM [$t(58) = -0.516$, $p > 0.05$] and between ASC and BEAM ($t(58) = -0.404$, $p > 0.05$] with the two SNRs combined. Similarly, there were no significant differences seen at each of the SNRs.

iii. Comparison of scores across SNRs

Further, the comparison of scores between the SNRs for each noise reduction algorithm was evaluated using two-way repeated measure ANOVA. The results showed that there was no significant difference in performance between SNRs for ADRO [$F(1, 34) = 0.074$, $p > 0.05$] and ASC [$F(1, 34) = 0.3$, $p > 0.05$]. Further, paired t-test showed that there was no significant difference in scores obtained using BEAM between the 2 SNRs [$t(11) = -1.476$, $p > 0.05$].

iv. Effect of microphone directionality on speech identification

In addition, the data were analysed to see the effect of microphone directionality on speech perception. This was done only for those using the BEAM algorithm since its functioning is based on the responses of the two microphones that are utilised. The data were compared between scores obtained from participants using CP180 and Freedom speech processor as the former utilised two omni directional microphones and the latter used one omni directional and one directional microphone. Independent t-test showed that there was no statistically significant difference in scores between the two processors using different microphone directionality [$t(22) = 0.263$, $p > 0.05$].

v. Comparison of scores across Speech processors

Further, the scores obtained were compared across speech processors (CP810, Freedom, Sprint/ESPrit 3G) using Kruskal-Wallis test. The comparison done for the 3 groups having 6 participants in each showed that there was no significant difference in scores across all the processors (Table 3).

Table 3: *Comparison of scores across Speech processors using Kruskal-Wallis test*

Condition	**Chi-Square**	**df**	**Asymp Sig**
Quiet	1.49	2	0.48
ADRO (+5 dB SNR)	0.71	2	0.70
ADRO (+10 dB SNR)	0.38	2	0.83
ASC (+5 dB SNR)	0.86	2	0.65
ASC (+10 dB SNR)	2.62	2	0.27
BEAM (+5 dB SNR)	0.32	1	0.57
BEAM (+10 dB SNR)	0.65	1	0.42

Test-retest reliability was ensured by carrying out the testing procedure on 2 participants. The scores did not vary when the test procedure was repeated after an interval of 2 months

Discussion

The impact of activating different noise reduction algorithms is evident from the *comparison of speech identification scores obtained in quiet and in the presence of noise*. The scores obtained in quiet with the participants using their default 'Everyday' settings, served as a baseline to compare the performance in the presence of noise with different noise reduction algorithms activated. The significant drop in scores in the presence of noise with all 3 noise reduction algorithms indicates that despite the use of these algorithms, their performance declined. As can be seen in Table 3, depending on the noise reduction algorithm used and the SNR, the mean decline in scored varied from 3.56 (14.24%) to 2.86 (11.44%).

The percentage drop in scores with the noise reduction algorithms is comparable or less than what has been reported to occur in individuals with normal hearing or hearing aid users at similar SNRs. Finitzo-Hieber and Tillman (1978) reported that in normal hearing children, at +6 dB SNR with a 0.4 reverberation condition, speech identification scores reduce by 21.2%. Likewise in their participants who were children with mild hearing loss, their speech identification scores with them wearing hearing aids dropped by 22%. Similarly, Johnson (2000) reported that in children with normal hearing, with an input signal of 40 dB SL, scores dropped by 15.6% to 12% depending on the age of the individual with the addition of a +13 dB SNR.

Thus, it can be inferred that the pre-processing strategies does not enable CI users to hear speech signals in a noisy situation similar to what they heard in quiet situation. However, the difficulty that they face in noisy situations is akin to or less than what normal hearing individuals or those with mild hearing loss would probably face.

Studies reported in literature regarding responses with ADRO indicate that the difference in scores without and with noise is far larger than what has been found in the current study. James et al. (2002) reported that the percentage drop in scores with ADRO was approximately 60% at +10 dB SNR and 20% at +15 dB SNR condition with the signal presented at 70 dB SPL. Such a large decline in scores was not evident in the present study. The variation in findings between the study by James et al. and the current study cannot be attributed to the type of processers / microphones used. While the former study used SPrint or ESPrit that utilise directional microphones, the present study used a combination of SPrint, ESPrit, Freedom and CP810 processors. The former two processors had directional microphones and the latter two had a combination of a directional and omni directional microphones. However, statistically no difference was noted between the different types of processors having different microphones. A possible reason for the variation could be the different stimulation rates used by James et al. that might have resulted in the poorer scores. In the present study, all the participants used a constant stimulation rate of 900 pps.

Dawson et al. (2004), similar to the present study, did not find a very large reduction in speech identification scores with the addition of noise. This reduction was less with the use of ADRO than with the use of a standard programme. However, besides presenting the test stimuli with and without noise, they also varied the input level of the signal. In quiet, the signal was presented at 50 dB SPL and in the presence of noise it was presented at 65 dB SPL. This makes it difficult to draw any direct conclusion about the difference in performance without and with noise. The study, however, did demonstrate that both in quiet and in a noisy situation, ADRO was effective in significantly improving speech identification scores.

In the current study, the *comparison of the scores across the 3 pre-processing strategies* (ADRO, ASC, & BEAM) resulted in there being no significant difference between the strategies. From Figures 1 and 2 it is apparent that this is reflected in the individual scores of the participants. This was evident in both the SNRs that were studied.

Studies demonstrating improvement in speech identification with ASC or BEAM in isolation is sparse. Brockmeyer and Potts (2011) noted that standard dual-port directional programme and ADRO had significantly poorer reception thresholds for sentences compared to ASC and BEAM. Such differences in the programmes were not observed in the present study. Variations in the procedure used for the presentation of the noise could have lead to the difference in findings in the present study and that got by Brockmeyer and Potts. The present study had an easier condition where the signal was presented from only one direction, while in the latter study the noise was presented from 8 speakers placed around the participants.

Thus, it is possible that only in listening conditions that are more difficult than what has been used in the present study, would there be a perceptual variation in the different pre-processing strategies. However, in a situation wherein noise and speech signals are generated from the front, which could often happen in a real life situation, no difference in performance would probably occur by varying the pre-processing strategy.

With *change in SNR* from +5 dB to +10 dB, no significant change in performance was seen in the current study with all 3 pre-processing strategies. This indicates that with higher noise levels (40 dB HL), the device enabled the individuals to continue perceiving speech in a similar manner as with a lower noise level (35 dB HL). However, in normal hearing children with almost similar increase in noise levels (+12 dB to +6 dB), a drop in performance by 11.4% has been reported by Finitzo-Hieber and Tillman (1978). In the present study, the maximum reduction in performance with increase in noise was just 4%.

The results of current study revealed that there was no effect of *microphone directionality*. Therefore, immaterial of whether the individual used directional, omni directional or a combination of a directional and omni directional microphones, the responses were similar. This lack of difference probably would occur only if both the signal and the noise arrive from the front of the listener.

As per a Cochlear company report, a significant mean improvement of 12% was noted with CP810 over Freedom processor (Clinical results with the new Cochlear Nucleus 5 system, 2010). However, the same was not reflected in the present study. Due to lack of details regarding how their study was carried out, reasons for the lack of consensus in the 2 studies cannot be specified.

Thus, from the present study it can be construed that when the noise and signals arise from the same direction, any pre-processing strategies (ADRO, ASC, & BEAM) can be utilised. Additionally, these pre-processing strategies perform in a similar manner at a lower (+5 dB) or higher (+10 dB) SNR. Directionality of the microphone also is not found to have an impact in such listening conditions.

Conclusion

From the findings of the study on 18 participants using CP810, Freedom, SPrint, and ESPrit 3G processors, it is evident that in the presence of noise their speech identification scores reduced when compared to their performance in quiet. This reduction was evident across all three pre-processing strategies (ADRO, ASC, & BEAM) at the 2 SNRs that were studied. The difficulty that the participants had in the presence of noise was similar to or less than what individuals with normal hearing or those with mild hearing loss were reported to have.

No significant difference in speech identification scores was seen between the 3 pre-processing strategies (ADRO, ASC, & BEAM) that were studied. Such results would probably occur only in situations where the noise and speech signals are generated from the front of the listener. Additionally, the scores did not vary across the 2 SNRs that were studied (+5 dB & +10 dB) for any of the pre-processing strategies that were studied. This highlights that cochlear implant users with pre-processing strategies activated, are not as adversely affected with the presence of noise as seen in individuals with normal hearing. This effect may occur only when the noise levels do not vary considerably. The study also found that when noise and speech are presented from the front of CI users, it did not matter whether they use processors with directional, omni directional or a combination of a directional and omni directional microphones.

From the findings of the study, recommendations can be made regarding the type of pre-processing strategy that should be used in typical listening situations, when the stimuli and noise arise from the front of the listener. The study also highlights the effect of microphone directionality on speech perception in the presence of noise.

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