Speech intelligibility in various noise conditions with the Nucleus® 5 CP810 Sound Processor

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Abstract

The Nucleus® 5 System Sound Processor (CP810, Cochlear™, Macquarie University, NSW, Australia) contains two omnidirectional microphones. They can be configured as a fixed directional microphone combination (called *Zoom*) or as an adaptive beamformer (called *Beam*), which adjusts the directivity continuously to maximally reduce the interfering noise. Initial evaluation studies with the CP810 had compared performance and usability of the new processor in comparison with the Freedom™ Sound Processor (Cochlear™) for speech in quiet and noise for a subset of the processing options. This study compares the two processing options suggested to be used in noisy environments, *Zoom* and *Beam*, for various sound field conditions using a standardized speech in noise matrix test (Oldenburg sentences test). Nine German-speaking subjects who previously had been using the Freedom speech processor and subsequently were upgraded to the CP810 device participated in this series of additional evaluation tests. The speech reception threshold (SRT for 50% speech intelligibility in noise) was determined using sentences presented *via* loudspeaker at 65 dB SPL in front of the listener and noise presented either via the same loudspeaker (S_0N_0) or at 90 degrees at either the ear with the sound processor (S_0N_{CI+}) or the opposite unaided ear (S_0N_{CI-}). The fourth noise condition consisted of three uncorrelated noise sources placed at 90, 180 and 270 degrees. The noise level was adjusted through an adaptive procedure to yield a signal to noise ratio where 50% of the words in the sentences were correctly understood.

In spatially separated speech and noise conditions both Zoom and Beam could improve the SRT significantly. For single noise sources, either ipsilateral or contralateral to the cochlear implant sound processor, average improvements with Beam of 12.9 and 7.9 dB in SRT were found. The average SRT of –8 dB for Beam in the diffuse noise condition (uncorrelated noise from both sides and back) is truly remarkable and comparable to the performance of normal hearing listeners in the same test environment. The static directivity (Zoom) option in the diffuse noise condition still provides a significant benefit of 5.9 dB in comparison with the standard omnidirectional microphone setting. These results indicate that CI recipients may improve their speech recognition in noisy environments significantly using these directional microphone-processing options.

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Introduction

Recipients of cochlear implants (CI) show remarkable speech recognition performance in quiet listening environments. Over the last 25 years, the results of word and sentence tests have continuously improved. A major factor of these improvements was attributed to more appropriate coding strategies and new sound processor technology as noted for example in a review paper[1]) where mean sentence recognition increased from 20% to 90% for the 22-electrode Nucleus system. Many authors did however notice that speech understanding in noisy and reverberant environments still presents a major challenge for cochlear implant users.[2,3] Progress in digital signal processing technology allowed implementations of powerful algorithms in hearing instruments and more recently in cochlear implant sound processors which can overcome some of the limitations of previous devices.[4] One possibility to reduce the impact of noise on speech recognition is the use of directional microphones, which pick up sound predominantly from one direction while attenuating signals from other directions. These devices, although with rather limited flexibility, have become quite popular in hearing instruments and were also used in early CI speech processors. With modern digital signal processing (DSP) systems, directionality can be adjusted in a more flexible way and combined with additional options.

The Nucleus® 5 System Sound Processor (CP810, Cochlear™, Macquarie University, NSW, Australia) contains two omnidirectional microphones. They can be configured as a fixed directional microphone combination (called *Zoom*) or as an adaptive beamformer (called *Beam*), which adjusts the directivity continuously to maximally reduce the interfering noise. Initial evaluation studies with the CP810 had compared performance and usability of the new processor in comparison with the Freedom™ Sound Processor (Cochlear™)[5] for speech in quiet and noise for a subset of the processing options.[6-8] The improvements, which were found with the new sound processor in comparison with the previous model, could be partly explained by differences in hardware and firmware.

The benefits of adaptive beamformer algorithms for CI recipients, which had been demonstrated in earlier studies,[9,10] were also found in these more recent investigations. However, the specific test setup in the laboratory sometimes tends to overestimate the real life benefit of a signal processing approach which is only one, although important, element of the whole system. Other solutions for CI recipients listening to speech in noisy environment have been proposed and evaluated as well[11] and a variety of alternative signal processing approaches have been studied or are under evaluation.[12]

The current study compares the two processing options suggested to be used in noisy environments, *Zoom* and *Beam*, for various sound field conditions using a standardized speech in noise matrix test [Oldenburg sentences test (OLSA)].[13] The two settings have been chosen as promising solutions for environmental noise situations where the target speech signal is spatially separated from the interfering noise sources. Unlike the clinical programming routine, which usually combines several processing options for optimal use in everyday listening situations,[14,15] the current study investigates each of the two directionality settings in isolation without additional preprocessing options. A similar test setup as the one used in the HearCom study of hearing instrument algorithm evaluation[12] with four loudspeakers for spatially coincident and separated sources for speech and noise was used. This procedure was adopted in an attempt to create somewhat realistic and challenging environmental situations for the CI recipients.

Materials and Methods

Study design

An intra-subject, repeated measures study design was conducted, where three settings of microphone directionality (omnidirectional, static directionality, adaptive directionality) were tested in four different signal and noise setups. All subjects served as their own control. Long-term use with one particular setting and only a short term trial with an unfamiliar setting could create an inherent bias in the performance and subjective impressions of individuals in favor of the setting they have had more experience with. Nonetheless, an equal trial period with each directionality setting is clinically not feasible, nor considered necessary for comparison purposes in view of the stability of the subjects baseline performance.

The study was conducted in accordance with the guidelines established in the Declaration of Helsinki. Ethics committee approval was obtained. All subjects had signed informed consent forms.

Statistics

Data analysis was performed using IBM SPSS Statistics version 22 (IBM Corp., Armonk, NY, USA). Analysis of variance was performed for the SRT values as the examined variables with microphone directivity and Signal and noise sources as independent factors. Group performance in the various listening conditions is presented graphically for individual subjects and as mean values. Differences between separated speech and noise sources conditions and the baseline condition with coincident speech and noise sources as well as differences between results with the Standard omnidirectional microphone setting and either Zoom or Beam settings were calculated and group mean and standard deviations are presented.

Subject selection

Subjects for this study were selected among experienced recipients of the CI24RE(CA) implant device (Cochlear™ Nucleus® Freedom™) who had previously participated in a multi-center study which evaluated the usability, comfort and reliability of the new Nucleus® 5 CP810 sound processor. Prior to receiving the CP810 sound processor these subjects had been using the Freedom™ speech processor for at least 3 months. Selection criteria for the current study included a minimum age of 18 years, proficiency in reading and writing in German language and capability to perform an adaptive speech recognition test in noise. Based on experience, subjects preferably should achieve a speech recognition score of at least 70% on OLSA in quiet in order to reliably reach an SRT in noise for Oldenburg sentences via an adaptive signal to noise ratio procedure.

Subjects were not considered candidates for the study if medical or psychological conditions would not allow performance of tests according to the study protocol or if additional handicaps would prevent participation in evaluations. Subjects with unrealistic expectations regarding the possible benefits, risks and limitations of the study procedures or inability to perform testing for speech recognition in noise on speech materials used were also excluded from participation.

Test setup

Speech recognition tests were performed via loudspeakers (Genelec Active Monitor Model 1029A, Genelec Inc., Natick, MA, USA) connected to a RME soundcard with 24-bit resolution (Hammerfall DSP Multiface II) in a sound treated test room as illustrated in

Figure 1. The reverberation time was verified to be low (approximately 0.1 s for frequencies between 250 and 6000 Hz). Test stimuli for speech recognition tests performed in quiet were presented from a speaker positioned at 0^0 azimuth, 1.5 m away from the subject's head when seated. For speech recognition tests in noise, three loudspeakers, 90^0 apart, each 1.5 m away from the subjects head were used, with speech stimuli presented from 0^0 azimuth and the competing background noise either from the CI-side (S_0N_{+CI}) (90° for subjects with CI on the right ear, –90° for subjects with CI on the left ear) or the opposite side (S_0N_{-CI}). An approximated diffuse noise field condition was generated by presenting uncorrelated noise signals simultaneously from three loudspeakers at 90°, 180° and 270° (S_0N_{DNF}).

To measure speech intelligibility in noise OLSA[13,16] was used with an adaptive signal-to-noise ratio test procedure with the competing background noise level set at 65 dB SPL. The intensity level of each sentence was adaptively adjusted following each response, to obtain SRT, *i.e.*, the signal-to-noise ratio at which a 50% correct speech understanding score was achieved.[17] The competing noise was generated by 30 random overlays of the whole test material, resulting in a signal with low amplitude modulations having the same long term spectrum as the test sentences, thereby leading to a high accuracy and reproducibility and very steep discrimination functions for speech intelligibility in noise of about 17%/dB for normal hearing listeners. A recent study[18] showed that the reproducibility and slopes for subjects scoring at positive signal to noise ratios (which is typical for CI recipients) may be reduced.

Speech recognition tests

A CP810 sound processor was programmed for each subject using the current patient map without preprocessing options as standard condition (omnidirectional microphone setting). Two additional programs with identical map parameters were generated with static microphone directionality (Zoom) and adaptive beamformer directionality (Beam). During the speech test sessions each of the three-processor programs was selected by the experimenter *via* the remote assistant (CR110) according to a predefined randomization table. The subjects were blinded as to the selected program options.

To establish suitability for the speech in noise tests two lists of 20 words each were used for the open set Freiburg monosyllabic word (MSW) test[19] in quiet and two lists of 10 sentences for the Oldenburg sentence test in quiet. The presentation level for these tests was 70 dB SPL. For the sentence tests in noise the noise level was kept constant at 65 dB SPL and the speech level was adjusted through an adaptive procedure[17] to yield a signal to noise ratio where 50% of the words in the sentences were correctly understood. Up to 30 sentences were used for each trial.

Training

As the CI recipients were long-term users with more than one year of implant experience, performance with their current sound processor was assumed to be generally stable. It was however necessary for all subjects to undergo a training period for familiarization with the OLSA test material and procedure and to remove potential learning effects when performing this test. The learning effect and the necessity for training have been previously described.[20,21]

Randomizations

Given the ABC paradigm for this study, it was important to check for and possibly remove learning effects and any associated bias with the test order. Therefore, the sequence of microphone directionality and speech and noise test conditions were varied across subjects via pre-configured randomization lists.

Results

Characteristics of enrolled subjects: monosyllabic word recognition scores

Nine subjects who fulfilled the inclusion criteria were selected for this study (Table 1). Their age ranged from 30 to 72 years (average of 53 years) and the duration of implant use ranged from 22 to 84 months (average of 63 months). All subjects performed very well in quiet. The monosyllabic word recognition (Freiburg MSW test) ranged from 70 to 100% with a mean value of 92.5% and a standard deviation of 10.5%.

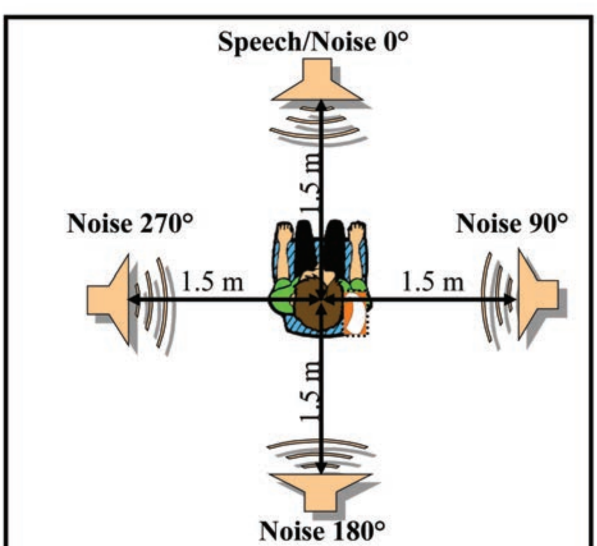


Figure 1. Test setup.

Oldenburg sentence recognition in quiet

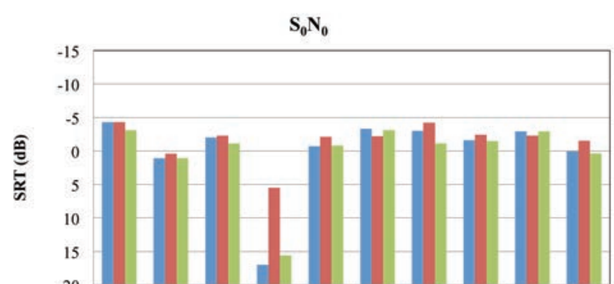
Sentence recognition in quiet as determined using the Oldenburg sentence test presented at 70 dB SPL is shown for all subjects (n=9) in Table 1. All subjects scored above 80% and therefore fulfilled the criterion to perform the adaptive speech in noise tests.

Evaluation of sentence recognition in noise

Oldenburg sentences test at S_0N_0

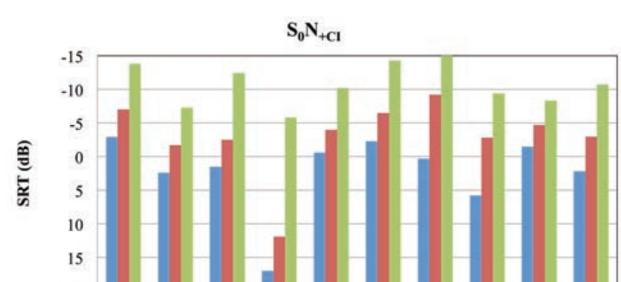
Results for OLSA sentences in competing background noise whereby speech and noise was presented coincidentally from one loudspeaker in front of the subject (S_0N_0) are shown in Figure 2. SRT's are expressed as the signal to noise ratio in dB at which 50% of the words in the sentences could be correctly repeated. Results for all subjects and mean results are displayed for the three microphone directionality settings (Standard, Zoom, Beam). Please note that a lower SRT value, *i.e.*, a worse signal to noise ratio, indicates improved performance of speech recognition in noise, which is emphasized by reversed ordinates of Figures 2-5. The mean values for all three microphone directionality settings were around 0 dB. The three settings did not result in statistically significant differences as confirmed by independent samples t-tests.

Subject S4 showed poor performance for Standard as well as Beam set-



	S1	S2	S3	S4	S5	S6	S7	S8	S9	mean
Standard	-4.3	1.1	-2	17	-0.7	-3.3	-3	-1.6	-2.9	0.03
Zoom	-4.3	0.4	-2.3	5.5	-2.1	-2.2	-4.2	-2.4	-2.3	-1.54
Beam	-3.1	1.1	-1.1	15.6	-0.8	-3.1	-1.1	-1.5	-2.9	0.34

Figure 2. Speech reception threshold (SRT), speech and noise from 0°, front (S_0N_0).



	S1	S2	S3	S4	S5	S6	S7	S8	S9	mean
Standard	-2.9	2.4	1.5	17	-0.6	-2.3	0.3	5.8	-1.5	2.19
Zoom	-7	-1.7	-2.5	11.9	-4	-6.5	-9.2	-2.8	-4.7	-2.94
Beam	-13.8	-7.3	-12.4	-5.8	-10.2	-14.3	-15.1	-9.4	-8.3	-10.73

Figure 3. Speech reception threshold (SRT), speech from 0o, noise from 90° or 270°, aided side (S_0N_{+CI}).

ting for coincident speech and noise presentation but much better performance for the Zoom setting. He also scored poorly with the Standard setting for spatially separated signal and noise (see below). In order to analyze this case in more detail the training sessions and randomization table were checked. In contrast to other subjects S4 had done 6 training sessions of 10 sentences each, reaching an average of 84% word recognition for the last two training lists. According to the randomization table he started the experiment with Standard setting (S_0N_0, S_0N_{+CI}), followed by Beam and Zoom settings and finally the two remaining conditions for Standard (S_0N_{-CI}, S_0N_{DSF}). Although a sequence effect cannot be ruled out completely in this case, it is remarkable that the two test conditions at the end of the experiment with coincident speech and noise sources yielded the same poor performance as the first two test conditions with the same speech and noise setup. As soon as the processing was changed from Standard omnidirectional to Beam the results were significantly improved and in the same order of magnitude as those of other subjects. Thus, it seems that this subject is very sensitive to noise contamination and enjoys excellent release from masking in spatially separated signal and noise conditions when activating the microphone directionality.

Oldenburg sentences test at S_0N_{+CI}

Results for OLSA sentences in competing background noise whereby speech was presented in front and noise from one loudspeaker at the side of the CI sound processor (S_0N_{+CI}) are shown in Figure 3. For all subjects a consistent performance improvement was observed when the microphone setting was changed from omnidirectional (Standard) to Zoom and Beam. Average SRT scores for the three settings were 2.19, –2.94 and –10.73. Differences between Standard and Zoom were not significant but differences between Standard and Beam and Zoom and Beam were highly significant ($P<0.0001$ and $P<0.005$). Please note the poorer performance for Standard omnidirectional microphone setting than for the coincident sound presentation in front (2.19 *versus* 0.03). Although this difference is statistically not significant it points to the more difficult listening condition with omnidirectional microphones when noise is presented close to ear with the CI sound processor.

Oldenburg sentences test at S_0N_{-CI}

Results for the test setup whereby speech was presented in front and noise from one loudspeaker at the side opposite to the CI sound processor (S_0N_{-CI}) are shown in Figure 4. For all subjects a consistent performance improvement was observed again when the microphone setting was changed from omnidirectional (Standard) to Zoom and Beam. Average SRT scores for the three settings were –2.58, –8.60 and –10.47. Differences between Standard and Zoom as well as Beam were significant ($P<0.05$) but differences between Zoom and Beam were not significant. The difference between the two conditions S_0N_{-CI} and S_0N_{+CI} was significant for Zoom ($P<0.05$) but not for the other two settings (Standard and Beam) although there is a trend for improved performance with the Standard setting for the noise presentation at the opposite ear which most likely is due to the head shadow.

Oldenburg sentences test at S_0N_{DSF}

Results for the diffuse sound field test setup whereby speech was presented in front and noise from three loudspeakers at both sides and at the back (S_0N_{DSF}) are shown in Figure 5. Again, it can be seen that all subjects performed consistently better when the microphone setting was changed from omnidirectional (Standard) to Zoom and Beam. Average SRT scores for the three settings were –1.32, –7.21 and –8.00. Differences between Standard and Zoom as well as Beam were significant ($P<0.05$) but differences between Zoom and Beam were not significant. The SRT's for the diffuse noise condition were worse than for the S_0N_{-CI} condition for the directional microphone settings, although statistically not significant. This effect was expected and can be explained by the principle of the beamformer algorithm, which predominantly reduces interfering signals from a single source.

Speech reception threshold differences for coincident *versus* spatially separated speech and noise sources

A univariate analysis of variance for the SRT values showed that both microphone directivity and signal and noise source setup were highly significant factors ($P<0.0001$). Their interaction was also highly significant ($P<0.001$).

Figure 6 shows the improvement in speech reception threshold for spatially separated speech and noise sources, relative to the reference condition with coincident speech and noise from 0°. Mean values of ($SRT_{S0N0} - SRT_{S0N+CI}$), ($SRT_{S0N0} - SRT_{S0N-CI}$), and ($SRT_{S0N0} - SRT_{S0NDSF}$) and standard deviations are shown. Mean values and standard deviations are listed in Table 2 (left).

The Standard (omnidirectional microphone) setting yields statistically significant differences between the ipsilateral and contralateral ($P<0.0001$) as well as the diffuse noise condition ($P<0.005$) whereas the differences between contralateral and diffuse noise conditions are not significant. The Zoom setting also yields statistically significant differences between the ipsilateral and contralateral ($P<0.001$) as well as the diffuse noise condition ($P<0.005$) whereas the differences between contralateral and diffuse noise conditions are not significant. None of the differences between noise conditions was significant for the Beam setting.

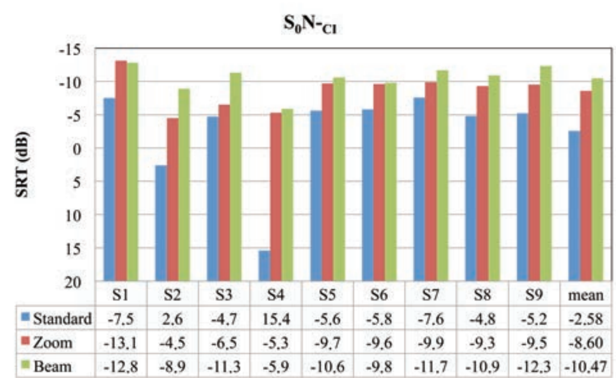


Figure 4. Speech reception threshold (SRT), speech from 0°, noise from 90° or 270°, unaided side, ($S0N_{-CI}$)

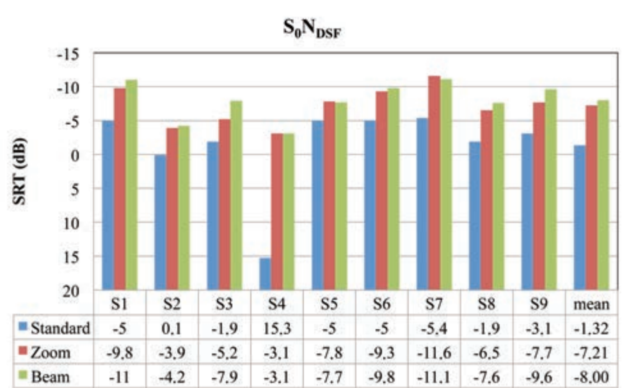


Figure 5. Speech reception threshold (SRT), speech from 0°, noise from 90°, 1800 and 270°, surround, diffuse sound field (S_0N_{DSF})

Performance benefits with directional microphone settings for spatially separated speech and noise

Differences between directional and omnidirectional microphone settings were analyzed for the sound presentations where speech and noise were generated by spatially separated sources.

Figure 7 and Table 2 (right) summarize the results. Improvements for the Zoom relative to the Standard setting ranged from 5.13 to 6.02 dB for the three noise conditions whereas the improvements for the Beam relative to the Standard setting ranged from 6.68 to 12.92 dB. Beam was most effective in the ipsilateral noise condition S_0N_{+CI} and outperformed Zoom significantly by more than 7.5 dB ($P<0.001$). For

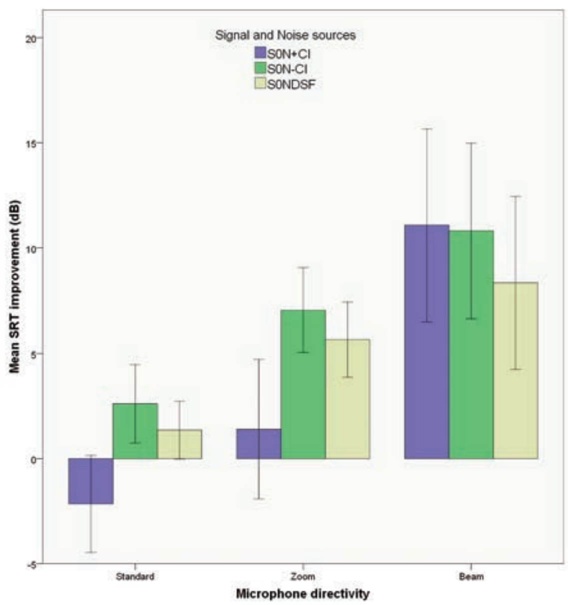


Figure 6. Improvement in speech reception threshold (SRT) for spatially separated speech and noise sources, relative to reference condition with coincident speech and noise from 0°. Mean values of (SRT_{S0N0} – SRTS0N+CI), ($SRTS_{0N0}$ – SRT_{S0N-CI}), and (SRT_{S0N0} – SRT_{S0NDSF}) and standard deviations are shown.

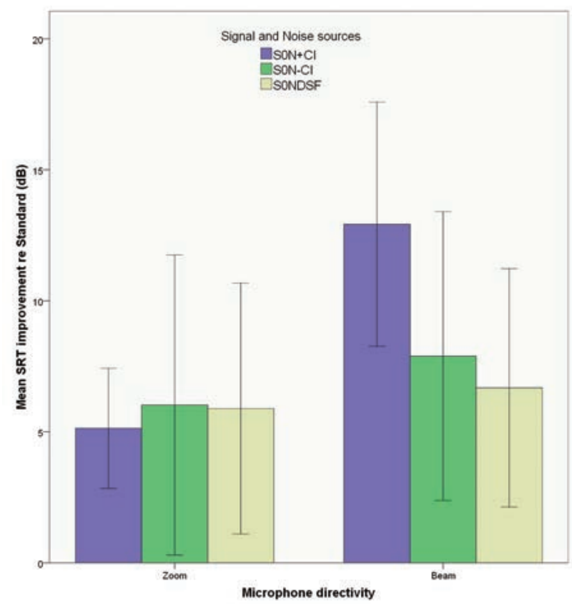


Figure 7. Improvement in speech reception threshold (SRT) for spatially separated speech and noise sources with microphone directionality (Zoom or Beam), relative to standard omnidirectional condition. Mean values of ($SRT_{Standard}$ – SRT_{Zoom}) and ($SRT_{Standard}$ – SRT_{Beam}) and standard deviations are shown.

Table 1. Subjects characteristics.

ID	Gender	Age (years)	CI use (months)	CI model	Ear	MSW (%)	OL-Q (%)
S1	M	43	65	CI24RE (CA)	R	100.0	100.0
S2	M	72	68	CI24RE (CA)	R	70.0	92.0
S3	M	59	63	CI24RE (CA)	R	92.5	98.0
S4	M	64	31	CI24RE (CA)	L	72.5	84.0
S5	M	30	22	CI24RE (CA)	L	95.0	88.0
S6	M	47	53	CI24RE (CA)	R	92.5	98.0
S7	M	55	70	CI24RE (CA)	R	95.0	100.0
S8	F	52	84	CI24RE (CA)	L	87.5	100.0
S9	M	53	61	CI24RE (CA)	L	95.0	98.0
Mean		53.0	63.0			92.5	98.0
St.dev.		12.2	19.5			10.5	5.9

CI, cochlear implant; MSW, Freiburg monosyllabic word; OOL-Q, Oldenburg sentences test in quiet; St.dev., standard deviation.

Table 2. Speech reception threshold differences for coincident *versus* spatially separated sources and for directionality options *versus* standard omnidirectional microphone setting. Mean values and standard deviations (in parentheses).

	SRT for coincident minus SRT for spatially separated sources			SRT for standard minus SRT for directionality	
	Standard	Zoom	Beam	Zoom	Beam
S_0N_{+CI}	–2.16 (2.31)	1.40 (3.32)	11.08 (4.58)	5.13 (2.29)	12.92 (4.66)
S_0N_{-CI}	2.61 (1.87)	7.06 (2.01)	10.81 (4.16)	6.02 (5.73)	7.89 (5.51)
S_0N_{DSF}	1.36 (1.37)	5.67 (1.79)	8.34 (4.10)	5.89 (4.79)	6.68 (4.55)

SRT, speech reception threshold.

the contralateral and diffuse noise conditions the differences between Beam and Zoom were not significant.

Discussion

The primary aim of this study was to evaluate possible SRT improvements through directional microphone settings for CI recipients using the CP810 sound processor. Unlike other studies where the performance with a new processor was compared to the performance with an earlier processor,[6,7,22] this study compared program settings within the same device. A restricted number of signal processing conditions was used in this study in contrast to other investigations which compared various combinations of preprocessing options.[23]

In some upgrade studies[6,8,15] of Freedom and CP810 processors significant performance differences were found for the Beam algorithm which were attributed to differences in microphone characteristics, matching accuracy and possible deviations from factory settings over time. The use of two omnidirectional microphones and firmware calibration in the CP810 should help to reduce hardware dependent variations and ensure similar conditions for every test subject. In the current study, a common test processor was used for all subjects in order to further reduce the influence of residual varying microphone characteristics. It should be noted, however, that the CP810 contains an electronic calibration table for each of the two microphones which provides superior matching of gain characteristics whereas the Freedom processor contains no integrated microphone adjustment mechanism.[7]

The speech intelligibility tests in quiet showed very good performance for the selected group of subjects. The mean value of 92.5% for monosyllabic word recognition is above some of the published results for the same test,[22] which may be explained by the special selection criteria of this study. All subjects were able to score above 80% word recognition for the Oldenburg sentence test in quiet after at least two training runs. An adaptive run for one experimental condition usually took about 5 or 6 min which makes the procedure feasible for clinical applications.[24] It should be mentioned that this study group might not be representative of arbitrarily selected CI recipients of a clinical population. The use of either static of adaptive microphone directionality was largely beneficial for spatially separated speech and noise sources but did not make a difference when speech and noise were presented from the same direction in front of the listener. The static directionality (Zoom) provided significantly less benefit for the ipsilateral noise condition than the adaptive setting (Beam). This was mainly due to the attenuation characteristic of the Zoom setting which reduces signals from 120^0 (or 240^0) ideally up to 15 dB and attenuates signals from 90^0 (or 180^0) by less than 5 dB as illustrated in polar plots of acoustic measurements.[6,15]

The large improvements with Beam processing compared with the omnidirectional microphone setting of 12.9 dB for the ipsilateral noise condition is comparable to the binaural performance of normal hearing subjects in similar setups.[12,25] The improvement with Beam for the contralateral and diffuse noise conditions of 7.9 and 6.7 dB respectively is also remarkable as it shows that the Beam algorithm is able to separate speech from noise in more complex configurations than for a single directional noise source.

The differences in dB SNR can be translated into performance differences in percent by taking into consideration the slope of the discrimination function. This slope has been determined for the Oldenburg sentence test in noise to be 17.1 %/dB for normal hearing subjects.[13] For test subjects who perform at less than 100% in quiet and at positive signal to noise ratios this slope value at the 50% point will generally be lower than 17.1%. Even with a conservative estimate of 10 %/dB for CI subjects[24] the SNR improvement with Beam versus the Standard omnidirectional setting condition translates to an average sentence intelligibility improvement of 70 to 100%. This means that a CI recipient who does not recognize a single word with the Standard setting would score up to 100% with the directionality setting activated for this specific speech and noise condition.

It should be pointed out, however, that these results are somewhat dependent on the room acoustic characteristics and the test setup. A more reverberant room such as the one which was used in the HearCom study[12] or smaller distances from the subject's head to the loudspeakers[8,14] might change the outcomes. Also, the Zoom and Beam conditions which were tested in the current study are usually combined with other preprocessing options in the clinical practice for everyday environment situations.[14] Studies which compared potential benefits and limitations of directional processing for hearing aids showed various results for speech recognition in noisy and reverberant environments depending on the amount and configuration of hearing loss.[4]

The use of speech spectrum shaped masking noise with very little temporal fluctuations was chosen for maximum SRT slopes.[16] Other types of masking noise[26,27] would eventually be better suited to reproduce real life difficulties of CI recipients. Thus, the benefits of Beam and Zoom processing demonstrated in the current study might be somewhat different for more realistic everyday listening environments. It should also be noted that the potential improvements of directional processing which are dependent on the room characteristics and the geometric distributions of target and interferer signals can only be obtained if the user actively selects the directionality option when appropriate and another option when the acoustic environment changes. Automatic processing selection was introduced with the successor devices of the CP810 processor, the CP910 and CP910.[28]

Conclusions

In spatially separated speech and noise conditions both Zoom and Beam can improve the SNR considerably.

For single noise sources either ipsilateral or contralateral to the CI sound processor average improvements with Beam of 12.9 and 7.9 dB in SRT were found.

The average SRT of –8 dB for Beam in the diffuse noise condition (uncorrelated noise from both sides and back) is truly remarkable and comparable to the performance of normal hearing listeners in the same test environment.

The static directivity (Zoom) option in the diffuse noise condition still provides a significant benefit of 5.9 dB in comparison with the standard omnidirectional microphone setting.

These results indicate that CI recipients may improve their speech recognition in noisy environments significantly using these directional microphone-processing options.

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