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An Influence of Directional Microphones on the Speech Intelligibility and Spatial Perception by Cochlear Implant Users

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The objective of the study is to assess the hearing performance of cochlear implant users in three device microphone configurations: omni-directional, directional, and beamformer (BEAMformer two-adaptive noise reduction system), in localization and speech perception tasks in dynamically changing listening environments. Seven cochlear implant users aided with Cochlear CM-24 devices with Freedom speech processor participated in the study. For the localization test in quiet and in background noise, subjects demonstrated significant differences between different microphone settings. Confusion matrices showed that in about 70% cases cochlear implant subjects correctly localized sounds within a horizontal angle of 30–40° (±1° loudspeaker apart from signal source). However localization in noise was less accurate as shown by a large number of considerable errors in localization in the confusion matrices. Average results indicated no significant difference between three microphone configurations. For speech presented from the front 3 dB SNR improvements in speech intelligibility in three subjects can be observed for beamforming system compared to directional and omni-directional microphone settings. The benefits of using different microphone settings in cochlear implant devices in dynamically changing listening conditions depend on the particular sound environment.

Keywords: localization, bilateral cochlear implants, adults, microphone, beamformer.

Abbreviations

SNR – Signal-to-Noise Ratio,
RMS – Root Mean Square,
AFC – Alternative Forced-Choice paradigm,
ACE – Advanced Combination Encoder,
DSP – Digital Signal Processing Algorithm,
FIR – Fixed Finite Impulse Response Filter,
SRT – Speech Reception Threshold.

1. Introduction

One of the most common complaints of both hearing aid and cochlear implant users relates to the perception of speech in noise. The simplest reason for inadequate conditions for proper speech perception is the poor speech signal-to-noise (SNR) ratio (Tyler *et al.*, 1983; Tyler, Kelsay, 1990; Hochberg *et al.*, 1992; Plomp, 1994; Holden *et al.*, 1995). In order to enhance the SNR, several different techniques have been implemented. The two most common rely on directional microphones and adaptive noise reduction systems. Both techniques use a different approach. Directional microphones take advantage of the spatial separation between noise and target by adjusting the point of maximum microphone sensitivity towards the target whereas noise reduction systems take advantage of differences in spectral and timing characteristics between target and noise arriving from multiple locations. Previous evaluations of different microphone configurations revealed considerable variations in methods and results depending on multiple variables, such as the number of interfering noise sources and characteristics of the stimuli (Holden *et al.*, 1995; Kochkin, 2000; Müller *et al.*, 2002; Qin, Oxenham, 2003; Strickney *et al.*, 2004; Kompis *et al.*, 2004; Wolf *et al.*, 2009; Brockmeyer, Potts, 2011), or noise reductions technique themselves (Spahr *et al.*, 2007, McCreery *et al.*, 2012; Wolfe *et al.*, 2012; Kokkinakis *et al.*, 2012; Magnusson *et al.*, 2013).

The microphone system type used as input for the cochlear implant is the main factor investigated in this study. The Freedom processor uses three microphone systems, including a forementioned directional system. There are two hardware microphones. The rear microphone is omni-directional and provides equal sensitivity regardless of the direction of incoming sound.

The omni-directional mode of operation is useful for overall monitoring of environmental sounds. A directional microphone located on top of the implant's external part forms a fixed cardioid pattern. A cardioidal directional characteristic favors frontal direction and diminishes signals received from rear (at 180°). This is obtained by using dual-port hardware, with anterior, and posterior microphone ports. There is an acoustical delay, by a few milliseconds, of the acoustic signal entering the posterior and anterior ports. Subtraction of these signals acting on the microphone diaphragm with opposite phases, cause reduced sensitivity of the directional microphone to sounds from the back. The microphone retains a high sensitivity to sounds from the front.

Freedom SmartSound Beam (beamformer) is the third most advanced mode of operation of the microphone systems. It takes advantage of the differences in spectral and timing characteristics between target and noise arriving from different locations. In this mode, the directional pattern is adaptive, using digital signal processing (DSP) algorithms and can change automatically depending on the sound in the environment. Beamforming mode uses both directional and omnidirectional microphones, employs two-channel summation of these signals, and performs second-stage digital processing in two channels. In contrast to a directional microphone, which cancels noise at 180 degrees, the beamforming mode cancels best when noise arrives at 90 degrees to the recipient. The digital algorithm requires time for processing of the signal, therefore adaptation can be as long as 1 s for speech-weighted noise, depending on the environment. In the beamforming systems, both front and rear microphones are activated simultaneously. The polar pattern changes adaptively using DSP algorithms and varies automatically depending on the environment. In the first stage of the signal, the pre-processing target and noise are spatially separated by filtering the output signal obtained from the omnidirectional microphone in the fixed finite impulse response FIR filter, and delaying the output signals from the directional microphone. In the second stage, speech and noise references are created. To create speech references, the output signal of the FIR filter is added to the delayed signal from the directional microphone. Finally, in the third stage, an adaptive noise cancellation algorithm is introduced (for a detailed description see Spriet *et al.*, 2007).

To our best knowledge there are not many publications which directly investigated effect of microphone type and settings on localization performance in cochlear implant users. For example Figueiro *et al.* (2001) examined localization performance, with expectation that localization abilities might be negatively impacted by the microphone configurations. His study explored effectiveness of the Audalion Beamforimng system in unilateral implanted pediatric population. Results showed, that subjects were unable to discriminate sound sources in none of four settings (regardless of the directionality) of the Audalion Beamforimng system.

The majority of conditions presented in the literature refer to listening where target and interfering noise remain in fixed positions (i.e., target arrives from the frontal sound source and interfering noise arises from the some lateral source), and originate from either single- or in some cases from multiple locations (e.g. Spriet *et al.*, 2007; Kokkinakis, Loizou, 2010). However in everyday situations, locations of target and interfering sound sources often vary independently, resulting in a dynamically changing listening environment.

The aim of the present study was to explore differences among omni-directional, directional and beamforming microphone configurations in laboratory settings that better represent real-life situations with a focus on individual performance of cochlear implant users. It should be noted that data in the literature typically presented averaged results therefore did not reflect advantages gained by individual patients. We intend to obtain the information that would supplement existing in literature knowledge about the speech perception in noise and provide new data regarding localization.

2. Materials and methods

2.1. Subjects

Seven adult subjects (six female and one male) implanted bilaterally with Nucleus CI-24M implant devices participated in this study. All individuals were post-lingual, severe-to profound sensorineural deaf. Subjects ranged from 27 to 68 years of age (mean

Table 1. The demographics of listeners and cochlear implant information.

Subject	Gender	Age [years]	Etiology of hearing impairment	Use of bilateral CIs [years]	Processor	Strategy	Number of channels	Pulse rate [Hz]
S1	F	58	Autoimmune Sensorineural Loss[a,b]	4	Freedom	SPEAK	18[a], 7[b]	250
S2	F	68	Unknown[a,b]	10	Freedom	ACE	22	1800
S3	F	50	Polio[a,b]	10	Freedom	ACE	20	2400
S4	F	62	Unknown[a,b]	7	Freedom	ACE	20	1800
S5	F	62	Hereditary[a,b]	2	Freedom	ACE	22	500[a], 250[b]
S6	M	27	Coggan's Syndrome[a,b]	5	Freedom	ACE	12	1800
S7	F	51	Unknown[a,b]	1	Freedom	ACE	22	2400

[a] right ear, [b] left ear.

of 54 years, SD = 13.5 years). A summary of subjects' demographic data is shown in Table 1. All subjects used the Freedom speech processor, which was programmed either with an Advanced Combination Encoder (ACE) speech processing strategy (subjects S2–S7) or a Spectral Peak (SPEAK) processing strategy (subject S1). Subjects were experienced implant users with at least 12 months of practice. All subjects received both implants during one surgical procedure (simultaneous implantation). All experimental procedures were approved by the University of Iowa Human Subject IRB. Patients' post implant performance in speech perception was evaluated by the consonant-nucleus consonant (CNC) monosyllabic words (Tilmann, Carhart, 1966) and City of University New York Sentences (CUNY) (Boothroyd *et al.*, 1985). Tests were conducted at each visit of implanted person commencing one month after implantation and continued to the beginning of the current experiment. Subjects demonstrated stabilized CNC and CUNY test results by the start of the current laboratory testing.

2.2. Measurement setup

Localization and speech perception tests were conducted in a 3×2.83×2 m (10′ × 9.3′ × 6.6′) soundproof booth (0.8-s average reverberation time) meeting ANSI standards. Stimuli were presented from an eight-loudspeaker array spanning a 108° arc symmetrically in front of the subject. The angular distance between loudspeakers was equal to 15.5°. Loudspeakers at angles of −54° to −8°, and of 8° to 54°, corresponded to locations on the left and right side of the median plane, respectively. Subjects were seated in the center at a distance of 1.5 m from the loudspeaker array.

A touch-screen monitor was placed in front of each subject to collect responses. The monitor was also used to display loudspeaker positions in localization tests, or a list of spondee words in speech recognition tests.

2.3. Experimental procedures

2.3.1. Localization tests

2.3.1.1. Stimuli for Localization Tests in Quiet and in Noise. A closed-set test was used. Sixteen everyday sounds representing four sound categories: warning and information signals (alarm, bell, Big Ben bell, and phone), vocalizations (child cry, bird1, bird2, duck, rooster, and dog), instruments (cello and guitar), and effects (breaking glass, knocking on wood, thunder and water noise) served as targets. For localization tests in quiet, target signals were presented nominally at 70 dB (C). For localization tests in noise, targets were presented at 60 dB (C) with the background noise at 50 dB (C). Noise and signal were cued simultaneously, all signals were prerecorded. Intensity of the targets and noise were kept at a constant level during testing. Background noise consisted of babble noise comprising 30 prerecorded sentences spoken by male and female talkers (5 male and 5 female talkers, 3 sentences per talker) (Tyler *et al.*, 2006). A non standardized test was used. Sentences were meaningful 5 to 7 words long.

2.3.1.2. Procedure. Signals were presented in six blocks. Each block consisted of all 16 everyday sounds, each played out only once within a block. Each of 16 different sound items was individually projected through one of the eight loudspeakers. In the localization test in quiet, each sound was presented from one of randomly selected loudspeakers. For localization tests in noise, in addition to a signal, babble noise was presented from two randomly selected loudspeakers, different from the loudspeaker used for signal presentation.

Positions of the eight loudspeakers in the array were displayed on the touch-screen monitor placed in front of a subject. Subject's task was to identify the loudspeaker from which the target originated. Before each stimulus presentation, subjects were instructed to focus on the center of the loudspeaker array, and to push a button at the touch screen monitor when ready. A forced-choice method was used in which subjects had to provide a response to each signal regardless to their ability to localize sound. Subjects were not restricted by time to give their response. No feedback was provided. Subjects were not familiar with the microphone configurations being tested. All participants were tested in a randomized order of listening conditions. A total of 96 trials (16×6) were presented to each subject. Errors in localization were determined by calculating the average root mean square (RMS) error in angular degrees.

2.3.2. Speech perception in noise

2.3.2.1. Stimuli. We have developed several realistic speech perception tests intended to assist in the evaluation of spatial hearing and bilateral devices (Tyler *et al.*, 2002a; 2002b; 2002c; 2007; Dunn *et al.*, 2004). In the 'cued' SRT test (Dunn *et al.*, 2008) spondee words were presented with background female-male babble noise. All signals were prerecorded.

Target spondees were introduced in a carrier phase "She saw the [target word]". The carrier phrase served as a localization cue, acting before the target spondee was presented. Listeners were given time to hear the 'cue' and move their head, in an attempt to replicate one of many possible realistic listening situations. Signal presentation began with the carrier phrase in quiet. Babble noise was turned on with a 600-ms delay and ended by the same time as the target spondee. In effect, the initial two words of the carrier phrase ("She saw") were presented in quiet, while the third word ("the") and the spondee were presented in babble noise. The spondee was introduced with 800 ms delay to the onset of the noise and with a delay of 400 ms to the end of the carrier phrase. The entire auditory cue (before spondee started) lasted for 963 ms and included the initial part of the carrier phrase (without noise), its final part (with noise), and noise alone before onset of the spondee. Such timing structure was used to allow the adaptive beamformer stabilize directionality, by pointing to the speech first, and then to include presentation of noise. The skim of signal timing stricture is presented in the Fig. 1.

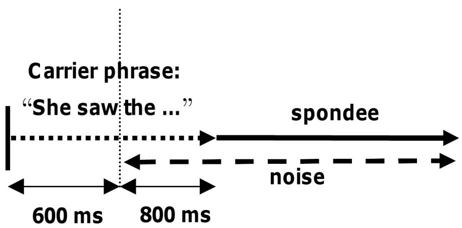

Fig. 1. Signal timing of target spondees presented in babble noise.

2.3.2.2. Procedure. A 12-alternative forced-choice (12-AFC) paradigm with 1-up/1-down adaptive procedure was used to determine the SNR threshold corresponding to the 50% correct identification. In this procedure a diagram of 12 spondees was presented to the subject on the monitor screen. The subject's task was to select the word played through the loudspeaker. Subject's response time was not limited. No feedback was given. Before each stimulus presentation subjects were instructed to direct their head to the center of the loudspeaker array, and to push a button at the touch screen monitor when ready. The subject was allowed to move their head during stimulus presentation.

In the adaptive procedure, the intensity of the spondee was kept constant at a level selected individually for each subject, to avoid ceiling and floor effects. The level of spondee presentation was nominally at 70 dB (C). In one instance it was lowered to 55 dB (subject S6) due to differences in subject's hearing dynamic range. Presentations level for the speech perception experiment was determined above the hearing threshold and for a subject's comfortable level of listening.

The initial noise level in the adaptive procedure was set 5 dB lower than the spondee level except for subjects S5 and S6 for whom the initial noise level was equal to the spondee level. In the adaptive procedure, the noise level was increased following each correct response, and was decreased following each incorrect response. The initial step size was 8 dB, and was lowered to 4 dB after the first level reversal. It was further lowered to 2 dB after the second reversal. Adaptive run was terminated after the 14th reversal. The threshold was calculated as the average level of the last 10 reversals. As the termination rule was based upon number of reversals, the total number of trials in the adaptive run varied between 25 and 35. In the single measurement session, five adaptive runs were conducted for each subject and each condition. The SNR corresponding to 50% correct word identification was calculated as the arithmetical mean of the values obtained in the last three runs.

2.3.2.3. Implant device programming. For the purpose of this study, a new cochlear implant map was created for each subject to avoid any uncontrolled advantages of previously used algorithm (Tyler *et al.*, 1986). The newly created maps were based on each subject's "everyday map" used on a daily basis. For each subject, individual changes were made in the programming parameters to the number of active electrodes and sampling rate. The coding strategy remained the same as in 'each subject's "everyday map". Using the comparison scale the subject indicates whether the current map settings was 'completely different', 'much different', 'bit different', or 'no change' from the "everyday map". Following the modification, the subject was asked to rate perceptual differences between his or her everyday map and the newly created map on a scale from 0 ('no change') to 100 ('completely different'). For the assessment each subject was interviewed by the audiologist with the same set of questions. Overall responses were rated by summation of all individual question scores and division through the maximum possible score. If the difference was <= 40%, the maps were considered similar, and further modifications were made. If the difference was >= 40% that map was applied in the experiment. Using research software, the speech processor was programmed specifically for omni-directional, directional, and beamformer modes of operation. During measurements the order of microphone conditions tested was randomized across the subjects. After specific microphone condition tested, microphone setting was always changed into another mode of operation (beamformer, directional or omni-diertcional). In all cases the speech processor microphone sensitivity was set to 6, and the input volume amplifier to values between 6 and 7.

3. Data analysis

We approached this study with regards to individual differences between subjects. Individuals may or may not benefit from highly directional microphones. With a limited number of subjects we chose to focus on individual results, and our statistical test reflect this approach. Data was mainly analyzed by descriptive statistics. Due to the small sample size, both paired t-tests and nonparametric Wilcoxon signed rank tests were carried out. As the results were consistent between the two tests the paired t-test results are reported. Analyses are carried out in SAS v9.2. In additional to individual analysis, group analyses were also performed, however due to the small sample size, this should be considered only as exploratory.

4. Results

4.1. Localization-in-quiet and localization-in-noise

The results for the localization tests are presented in Figs. 2 and 3. In both figures, stimulus positions are shown on the abscissa and subject's responses on the ordinate. Negative and positive values on both axes represent left and right sides of the loudspeaker arc respectively. Each column represents data for different device settings: omni-directional microphone, di-

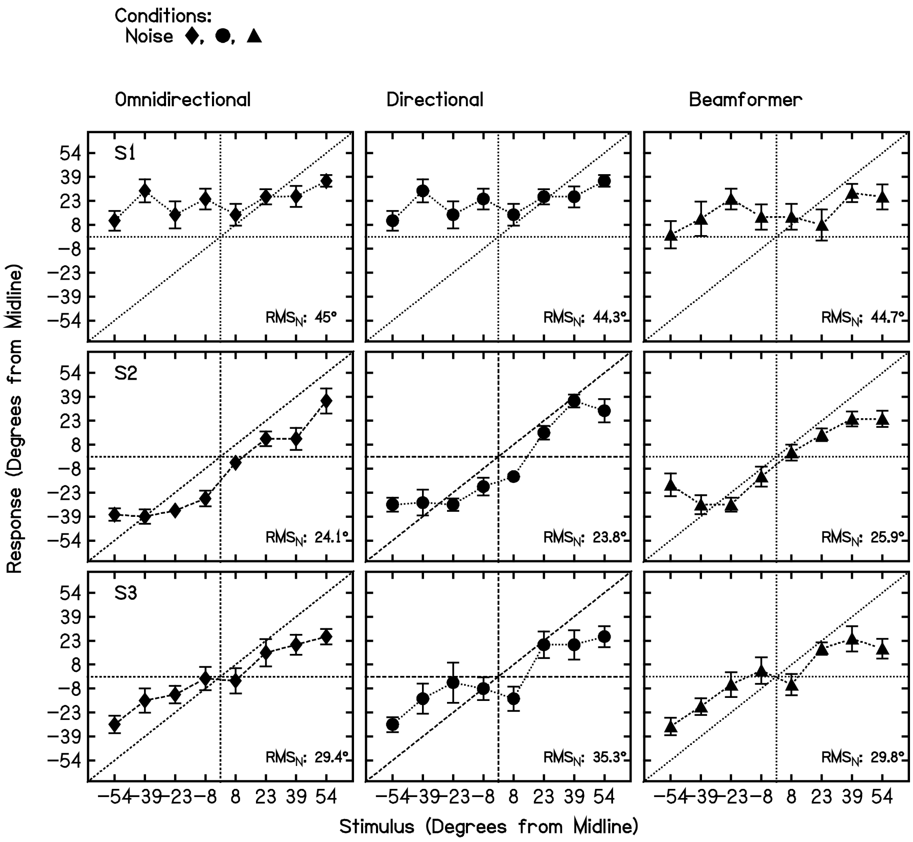

Fig. 2. Scattograms for the localization test in noise. Columns correspond to the three microphone configurations: omni-directional (first column), directional (second column), beamformer (third column). Rows represent data obtained from subjects S1–S3. The abscissa indicates the stimulus locations in the range from −53° to −8° (left side) and from +8° to +53° (right side). Subjects' responses are shown on the ordinate. Diagonal lines represent correct performance.

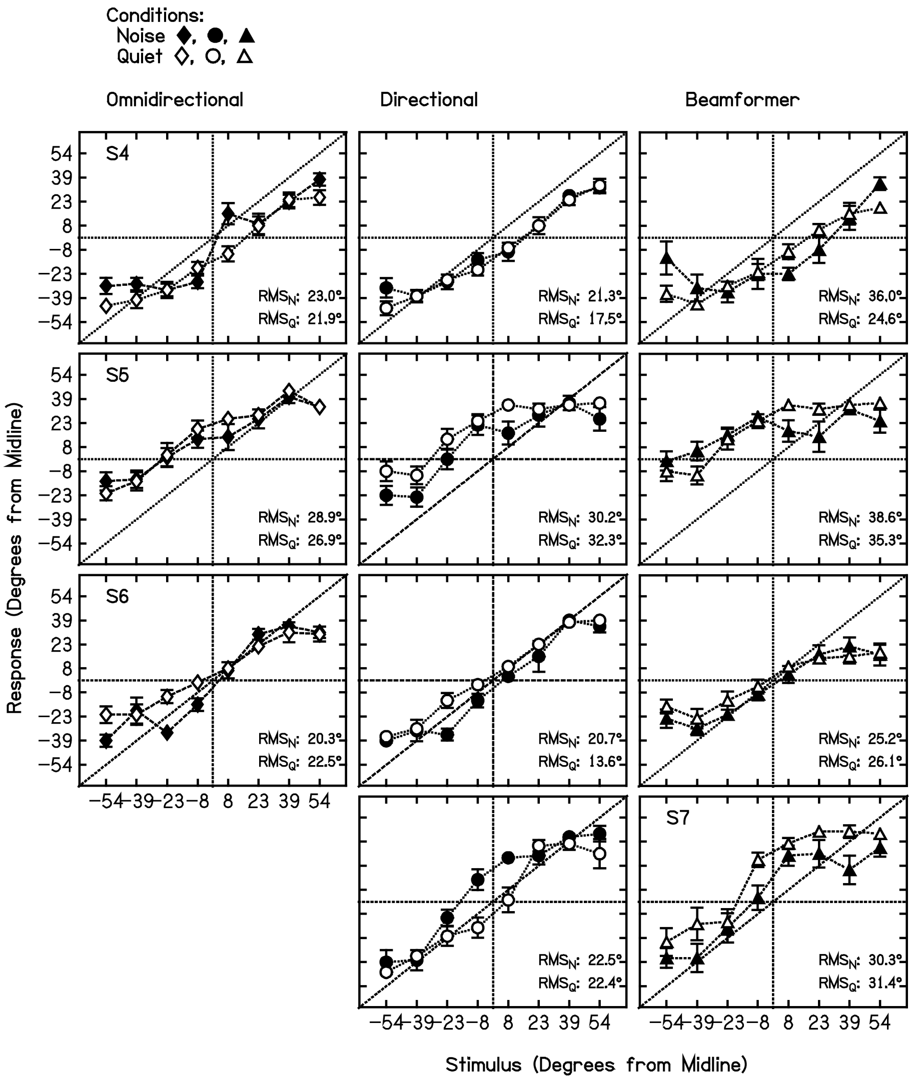

Fig. 3. Scattograms for the localization test in noise and quiet for subjects S4–S7. Closed symbols refer to the condition tested in noise, open symbols refers to the condition tested in quiet. Other details as in Fig. 2.

rectional microphone, and beamforming system. Each row represents data of individual subjects. Numbers displayed inside panels represent RMS errors averaged over all subject's responses. Closed symbols in Fig. 2 refer to listening condition in noise (subjects S1–S3), whereas closed and open symbols in Fig. 3 refer to listening conditions in noise and in quiet respectively (subjects S4–S7). Due to time constrains, subject S7 did not complete the task for the omni-directional microphone, and the localization test in quiet was not performed by subjects S1–S3.

For the localization test in quiet and localization test with the background noise, three general patterns of responses, accompanied by differences in subjects' localization performance, were observed. Four subjects (S2, S3, S6, and S7) accurately localized towards the center of the loudspeaker array, while two other subjects (S1 and S5) generally localized towards the side. Subject S1 localized all sounds on the right side of the median plane. It should be noted that for each condition tested, both cochlear implants devices were balanced for loudness.

Subjects S5's asymmetric pattern of localization was different. Subject S5 localized sounds presented on the left side as originating from loudspeakers located approx. two positions further to the right. However, sounds presented on the right side were localized with suitable accuracy. Among all seven subjects only subject S4 displayed a wide localization pattern, preserving positions of loudspeakers with a small standard deviation of responses (see Fig. 3, upper panels).

For localization tested in noise, a significance level of 0.05 did not reveal statistically significant differences between directional and omni-directional microphones ($T = 0.755$, $p = 0.4902$) in average group results. Group averages for directional and omni-directional microphone settings vs. beamforming settings were also not statistically significant at a level of 0.05 (beamforming vs. omni-directional microphone: $T = 2.23$, $p = 0.076$; confidence interval: -0.7454, 10.58; and beamforming vs. directional microphone: $T = -1.89$; $p = 0.108$; confidence interval: -10.63, 1.37. Subject S7 did not participate in tests with omni-directional microphone settings therefore his data was excluded from statistic analysis. Lack of the statistical significance was most likely due to the restricted sample size (number of subjects).

For subjects S3, S4, S5, and S6, differences were observed in individual results between the beamforming and both omni-directional and directional microphone settings. In all conditions tested the differences were associated with the poorer performance under beamformer system. Significant differences were also observed for the beamforming system and directional microphone for subject S7, with the directional microphone advantageous over the beamforming microphone settings.

In summary, data in Figs. 2 and 3 reveal that neither localization test in quiet nor localization test in noise showed an advantage of the beamforming microphone settings over directional or the omni-directional microphone. However, the background noise led to changes in response patterns, indicating a less accurate identification of the loudspeaker from which the signal originated. Most subjects displayed constant errors in localization, seen as a shift of localization towards the median plane or one side. A summary of localization errors shown in each panel of Figs. 2 and 3 is given in Table 2. Numbers at the bottom rows represent average errors calculated over all subjects and all eight loudspeaker positions (first row), and average errors calculated for subjects S4–S7 and all eight loudspeaker positions (second row) respectively. The values listed in Table 2 related to performance of individual subjects show that smaller RMS errors represent better localization ability occurred in omni-directional and directional microphone settings. For signal and noise presented from $\pm 54°$ angles in front of subjects, neither use of the directional microphone nor the beamformer system improved localization.

For further illustration of localization ability in quiet and in noise for each microphone condition tested, confusion matrices (Tables 3 and 4) averaged over all subjects was calculated. Confusion matrices provide more detailed information than RMS errors shown in Figs. 2 and 3 as they reveal possible asymmetry in the directional response patterns provided by subjects. Columns in Tables 3 and 4 refer to loudspeakers used for signal presentation, whereas rows correspond to subjects' responses pointing to a particular loudspeaker. Numbers 1 to 4 and 5 to 8 indicate loudspeakers positioned on the left and right side, respectively. Percent of correct direction identification averaged over responses obtained from seven (localization in background noise) or four subjects (localization in quiet) for different microphone settings – omnidirectional, directional, and beamforming – are shown in the upper, middle, and lower parts of Tables 3 and 4, respectively. Table diagonals highlighted by solid boxes indicate perfect localization performance. As such perfect localization can be considered as too demanding for cochlear implant users, more relaxed criterion was also used, in which selection of a loudspeaker that was one loudspeaker apart from that generating the signal was still considered a correct identification. Applying relaxed criterion includes data on main diagonal and on $\pm 1°$ side diagonals into the count of percent correct responses, which is shown in Tables 3 and 4 highlighted by dashed boxes, and referred later in the text as an extended diagonal.

Table 2. Individual and average localization errors (RMS in degrees) for different microphone settings NA-not applicable.

	Conditions in quiet			Conditions in noise		
Subject	Omni-directional	Directional	Beamformer	Omni-directional	Directional	Beamformer
S1	NA	NA	NA	45°	44.3°	44.7°
S2	NA	NA	NA	24.1°	23.8°	25.9°
S3	NA	NA	NA	29.4°	35.3°	29.8°
S4	21.9°	17.5°	24.6°	23.0°	21.3°	36.0°
S5	26.9°	32.3°	35.3°	28.9°	30.2°	38.6°
S6	22.5°	13.6°	26.1°	20.3°	20.7°	25.2°
S7	NA	22.4°	31.4°	NA	22.5°	30.3°
Average across all subjects	23.8	21.5	29.4	28.5	28.3	32.9
Average across subjects (S4–S7)	23.8	21.5	29.4	24.1	23.7	32.5

Table 3. Confusion matrix for localization in noise.

		Stimulus identification [%]							
		Position of loudspeaker							
		1	2	3	4	5	6	7	8
		Omni-directional							
Subject response	1	19.4	8.3	9.7	4.2	2.7		1.4	1.4
	2	22.2	30.6	23.6	15.3	1.4	2.8	1.4	1.4
	3	30.6	25.0	25.0	27.8	12.3	6.9	2.8	1.4
	4	8.3	8.3	13.9	12.5	27.4	8.3	1.4	1.4
	5	8.3	11.1	15.3	15.3	20.6	16.7	12.5	8.3
	6	5.6	5.6	4.2	11.1	17.8	26.4	31.9	30.6
	7	5.6	8.3	6.9	11.1	13.7	33.3	33.3	34.7
	8		2.8	1.4	2.8	4.1	5.6	15.3	20.8
		Directional							
Subject response	1	17.9	16.7	14.3	3.6	4.8	3.6	1.2	1.2
	2	39.3	33.3	17.8	7.1	5.9			3.6
	3	17.9	21.4	27.4	26.2	9.5	4.8	2.4	
	4	13.1	8.3	14.3	23.8	34.5	5.9	1.2	2.4
	5	2.4	3.6	10.7	13.1	19.0	15.5	8.3	7.1
	6	7.1	3.6	4.8	14.3	13.1	28.6	30.9	22.6
	7	2.4	7.2	4.8	9.5	13.1	33.3	34.5	45.2
	8		5.9	5.9	2.4		8.3	21.4	17.9
		Beamformer							
Subject response	1	17.9	17.9	7.1	4.8		2.4	1.2	
	2	17.9	27.4	22.6	9.5	8.3	5.9	1.2	1.2
	3	25.0	20.2	25.0	20.2	14.3	8.3	2.4	5.9
	4	13.1	10.7	9.5	13.1	22.6	10.7	10.7	7.1
	5	10.7	5.9	9.5	16.7	17.9	21.4	15.5	10.7
	6	7.1	8.3	15.5	17.9	17.9	20.2	28.6	28.6
	7	8.3	4.8	8.3	15.5	11.9	21.4	23.8	30.9
	8		4.8	2.4	2.4	7.1	9.5	16.7	15.5

Table 4. Confusion matrix for localization in quiet.

		Stimulus identification [%]							
		Position of loudspeaker							
		1	2	3	4	5	6	7	8
		Omni-directional							
Subject response	1	19.4	13.2	5.1			2.8		
	2	30.6	36.8	17.9	5.6	5.6			
	3	25.0	23.7	33.3	16.7	2.8			
	4	19.4	10.5	15.4	33.3	19.4		8.3	
	5	5.6	7.9	23.1	25.0	38.9	30.6	5.6	19.4
	6		7.9		11.1	25.0	47.2	27.8	30.6
	7			2.6	5.6	8.3	19.4	30.6	38.9
	8			2.6	2.8			27.8	11.1
		Directional							
Subject response	1	29.2	20.8						
	2	35.4	31.3	18.8	6.3				
	3	14.6	31.3	27.1	10.4	2.1			
	4	8.3	6.3	22.9	31.3	18.8	4.2		
	5	8.3	6.3	14.6	27.1	31.3	29.2	6.3	6.3
	6	4.2	4.2	10.4	8.3	20.8	35.4	31.3	18.8
	7			6.3	12.5	25.0	25.0	43.8	47.9
	8				4.2	2.1	6.3	18.8	27.1
		Beamformer							
Subject response	1	18.8	16.7	6.3	2.1			2.1	
	2	27.1	33.3	12.5	8.3	2.1			
	3	16.7	20.8	31.3	10.4	6.3	2.1	4.2	2.1
	4	16.7	6.3	18.8	16.7	14.6	4.2		
	5	16.7	8.3	6.3	27.1	25.0	29.2	14.6	18.8
	6	2.1	6.3	16.7	18.8	12.5	18.8	43.8	37.5
	7	2.1	8.3	6.3	12.5	33.3	27.1	14.6	22.9
	8			2.1	4.2	6.3	18.8	20.8	18.8

A summary of the data from Tables 3 and 4 is given in Table 5, in which a total percent identification on the main and extended diagonals is listed. For the omni-directional microphone in noise conditions, only 23.6% of subjects' responses correspond to the main diagonal; however, 62.7% of subjects' responses correspond to the extended diagonal, indicating that subjects' performance is not perfect, but satisfactory from a practical point of view. There is a slight improvement of correct identification with the directional microphone (25.3% and 68.2%), but no improvement for the beamforming system (20.1% and 55.7%). Results in Table 4 reveal better subject performance for localization tasks in quiet than in noise. Correct identification is increased to 30–40% in many instances (for data on main diagonal see Table 4). There is also a smaller number of responses selecting loudspeakers remote from that which were used to generate signals. In quiet conditions, the beamforming system did not lead to significant improvement in localization. The percent of substantial identification

Table 5. Summary of correct identification for Omni-directional, Directional and Beamformer microphone configurations in quiet and noise.

Microphone configu-ration	Percent correct identification on diagonal			
	Quiet		Noise	
	Main diagonal	Extended diagonal	Main diagonal	Extended diagonal
Omni-directional	31.4	72.7	23.6	62.7
Directional	32.0	76.8	25.3	68.2
Beamformer	22.1	60.2	20.1	55.7

errors for the beamformer system (see lower part of Table 4) is larger in comparison to the omni-directional and directional microphones settings (see middle and upper part of Table 4). The summary given in Table 5 indicates 31.4% and 32.0% correct identification for omni-directional and directional microphones, respectively, and only 22.1% in beamforming condition. Applying the extended diagonal criterion leads to correct identification in quiet of 72–76% for omni-directional and directional microphones, and of 60.2% for beamforming system. These numbers represent 10–15% higher identification level than correct identification in noise.

Data in Figs. 2 and 3 also show that the percentage of correct identification is not related to the positions of loudspeakers spanning the arc of ±54°, as percent of correct responses is similar along main diagonals of confusion matrices. Thus, within a frontal angle of approx. one-third of the circle, localization of sound sources in quiet and with background noise is comparably effective.

4.2. Speech perception in background noise

Individual results obtained from subjects S1, S2, and S4–S6 for speech perception in background noise are shown in Table 6 (due to time constraints test was not performed by subjects S3 and S7). First, it should be noted that level of the 50% spondees identification differs in SNRs among subjects by about 20 dB (see column "Average" in Table 6). For two subjects S2 and S6 comparison of the three microphones settings reveals slight improvement in the SNR for beamforming system over directional or omni-directional microphone e.g. −2.8 and −5.3 for beamformnig vs. directional microphone comparisons, and −2.8 and −4.7 for beamforming vs. omni-directional microphone comparisons. Although individual differences across 368 subjects, test conditions, and microphone settings were observed, these findings don't reflect on averages calculated for the entire subject population (bottom row of Table 6).

Table 6. Individual and average SNR in (dB) for different microphone settings.

	Conditions		
Subject	Omni-directional	Directional	Beamformer
S1	2.3	−0.1	0.6
S2	−2.1	−2.1	−4.9
S4	−2.2	−1.5	−1.7
S5	−11.3	−12.1	−9.5
S6	−17.5	−18.1	−22.8
Average	−7	−6.8	−7.9

5. Discussion

Our study investigated benefits of three microphone configurations – omni-directional microphone, directional microphone and SmartSound Beam (beamforming system) – on localization abilities in quiet and in noise, and on speech perception in noise in dynamically changing listening environments. With ongoing changes in the spatial location between target and noise, the potential advantages of different systems are not straightforward. As an example, confusion matrices for localization test in noise (see Tables 3 and 4) show that replacing an omni-directional with a directional microphone and further with a beamforming system is not followed by improvement in localization. However, for conditions away from the diagonal, some improvement is seen for the directional microphone as compared to the omni-directional microphone, meaning that smaller number of large errors in identification occurs. While some subjects demonstrated improvements in speech perception in noise, a group-wide comparison of the three microphone settings revealed slight or no improvements for the beamforming system or directional microphone over the omni-directional microphone. Nevertheless some differences within and among subjects were observed. For example the use of the beamforming system, compared to the directional microphone, resulted in improvements for subjects S2 and S6 (−2.8 dB and −4.7 dB), but not for subjects S1, S4, and S5.

For a person listening with two ears (without hearing aids), the binaural system receives and analyzes signals arriving from different locations. Based on the interaural level timing and spectral differences, the central auditory system does its best to locate sounds and hear speech in the presence of multiple background noises distortions introduced by hearing loss can disrupt some of these cues. As hearing aids and cochlear implants are particularly adept at improving performance in quiet, the signal processing involved has the potential to distort the normal bilateral level, timing, and spectral differences. The cues normally used by the central auditory system to identify sound sources and spatially separate speech in noise are attenuated by the directional characteristic of the devices.

The beamforming system and the directional microphone can alter localization cues causing poorer localization abilities. Our study shows that in localization tasks the beamforming system did not improve subjects' performance compared to directional and omni-directional microphones. The beamforming system should however perform better in speech intelligibility tests in noise, as it is designed to minimize masker effects. The potential advantages for beamforming system over omni-directional and directional microphones are not straightforward.

Our data shows slight improvements in the SNR in speech intelligibly by some individuals for beamforming settings, which is consistent with data presented in the literature. For example Wouters and Van Den Berghe (2001) obtained an average improvement of 10 dB in the SRT for speech-weighted noise and ICRA noise (ICRA, Reference Note 1), given a single noise source. For the SRT thresholds at noise level of 55 dB SPL, Spriet *et al.* (2007) reported 7.5 dB improvement for speech-weighed and babble noise presented from single source, and correspondingly 2 dB and 5 dB improvement for multiple noise sources. For speech-weighed noise and babble noise at a 65 dB SPL, the beamforming system, was even more effective, and led to an improvement of 13–16 dB in the SRT thresholds for a single-noise source and 7–11 dB for multiple-noise sources. Spriet *et al.* (2007) concluded that the beamforming system improved speech intelligibility in all conditions in noise, with the most visible improvement for a single-noise source. This result was in agreement with data presented by Hu and Loizou (2007). Using different types of noise, Hu and Loizou (2007) concluded that speech enhancement algorithms appeared to depend on the temporal and spectral characteristic of noise, particularly in low SNR conditions. Most recently, Kokkinakis and Loizou reported average 20% improvement in intelligibility of words for multiple-noise sources, and 30% improvement for single-noise source, for an algorithm based on four microphones coupled binaurally.

As shown, benefits for particular microphone systems are strongly dependent on the particular stimulus conditions and timing relationships between the targets and noises. An important factor is that time is required by the beamforming signal processing, to determine the location of the signal and noise. Errors can be made in this decision, and time is required to change the directionality characteristics of the beamformer.

It should be also noted that the beamforming system is expected to work best for side (90°) and back (180°) positions of background noise. Tests conducted in our laboratory setup, in which all loudspeakers were located in front of subjects within ±54° of median plane, showed that the beamforming system did not display a meaningful advantage for localization test over the omni-directional microphone. This result although not statistically significant may suggest that in all conditions in which the background noise is diffused, such as in reverberant spaces (when noise is not coming from the back), the advantage of beamforming system may be limited.

An important limitation of our study is the restricted number of subjects involved. We mainly explored individual differences, and also highlighted the importance of directional microphones for the particular task at hand. There is a large variety of real listening situations. Our study emphasizes the importance of considering dynamically changing listening tasks when evaluating different directional microphone characteristics.

6. Conclusions

- Confusion matrices showed that cochlear implants displayed sufficient localization ability in quiet, as in about 70% cases subjects correctly localized sounds within a horizontal angle of 30–40° (±1° loudspeaker apart from signal source).
- Localization in noise was less accurate than in quiet, as shown by a large number of considerable errors in localization in the confusion matrices.
- The importance of localization (and the RMS error scores reported) depends on the listening situation. However the average result (Table 6) indicates the lack of statistically significant differences between three microphone configurations.
- For speech presented from a frontal angle of ±54° slight SNR improvements in speech intelligibility in some individuals can be observed for beamforming system compared to directional and omni-directional microphone settings.

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