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Speech understanding in background noise with the two-microphone adaptive beamformer BEAM^{TM} in the Nucleus Freedom^{TM} cochlear implant system [1]

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Speech Understanding in Background Noise with the Two-Microphone Adaptive Beamformer BEAM™ in the Nucleus Freedom™ Cochlear Implant System

Ann Spriet, Lieselot Van Deun, Kyriaky Eftaxiadis, Johan Laneau, Marc Moonen, Bas van Dijk, Astrid van Wieringen, and Jan Wouters

Objective: **This paper evaluates the benefit of the two-microphone adaptive beamformer BEAM™ in the Nucleus Freedom™ cochlear implant (CI) system for speech understanding in background noise by CI users.**

Design: **A double-blind evaluation of the two-microphone adaptive beamformer BEAM and a hardware directional microphone was carried out with five adult Nucleus CI users. The test procedure consisted of a pre- and post-test in the lab and a 2-wk trial period at home. In the pre- and post-test, the speech reception threshold (SRT) with sentences and the percentage correct phoneme scores for CVC words were measured in quiet and background noise at different signal-to-noise ratios. Performance was assessed for two different noise configurations (with a single noise source and with three noise sources) and two different noise materials (stationary speech-weighted noise and multitalker babble). During the 2-wk trial period at home, the CI users evaluated the noise reduction performance in different listening conditions by means of the SSQ questionnaire. In addition to the perceptual evaluation, the noise reduction performance of the beamformer was measured physically as a function of the direction of the noise source.**

Results: **Significant improvements of both the SRT in noise (average improvement of 5–16 dB) and the percentage correct phoneme scores (average improvement of 10–41%) were observed with BEAM compared to the standard hardware directional microphone. In addition, the SSQ questionnaire and subjective evaluation in controlled and real-life scenarios suggested a possible preference for the beamformer in noisy environments.**

Conclusions: **The evaluation demonstrates that the adaptive noise reduction algorithm BEAM in the Nucleus Freedom CI-system may significantly increase the speech perception by cochlear implantees in noisy listening conditions. This is the first monolateral (adaptive) noise reduction strategy actually implemented in a mainstream commercial CI.**

(Ear & Hearing 2007;28;62–72)

Although most cochlear implant (CI) users today achieve remarkably good speech understanding in quiet, they generally experience severe performance degradation in noisy acoustical environments. The capability of an individual listener to understand speech in noise is often measured by the speech-reception threshold (SRT). The SRT is defined as the signal-to-noise ratio (SNR) where 50% of the speech is correctly understood. While people with normal hearing are able to understand 50% of the speech in a noisy environment with an SNR as low as −5 dB (Plomp & Mimpen, 1979; Versfeld, Daalder, Festen & Houtgast, 2000; Wouters, Damman & Bosman, 1994), cochlear implantees have an SRT between 5 dB and 15 dB and hence, require an SNR that is 10 dB to 25 dB higher (Hochberg, Boothroyd, Weiss & Hellman, 1992; Parkinson, Parkinson, Tyler, Lowder & Gantz, 1998; Wouters and Vanden Berghe, 2001).

The fact that CI users have great difficulty understanding speech in noise is partly due to the loss of spectral resolution. Indeed, it has been shown that an increase of the effective number of spectral channels improves the speech reception in noise of CI users (Fu, Shannon &Wang, 1998; Friesen, Shannon, Baskent & Wang, 2001). However, due to current spread (i.e., channel interaction), there is a practical limit to the effective number of independent stimulation channels in any intracochlear electrode array. This limit is, in general, lower than the optimal number of spectral channels for speech understanding (in noise). Hence, increasing the number of electrodes in CIs will not improve speech understanding in noise (Friesen et al., 2001). In multitalker background noise, the performance may be further improved by combining electrical stimulation with residual low-frequency acoustic hearing (Turner, Gantz, Vidal, Behrens & Henry, 2004). In stationary noise, however, no improvement has been observed.

An effective way to improve the ability to understand speech in background noise, is to incorporate a

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noise reduction algorithm as a preprocessor to the CI's speech processor. In the last decade, a number of noise reduction strategies have been implemented and tested for CIs (Weiss, 1993; Van Hoesel & Clark, 1995; Hamacher, Doering, Mauer, Fleischmann & Hennecke, 1997; Kompis, Feuz, Valentini & Pelizzone, Reference Note 1). As an example, experiments by Van Hoesel and colleagues (1995) showed an average improvement in sentence perception scores of 31% for a single noise source at 90° with Nucleus-22 cochlear implantees using a bilateral two-microphone adaptive beamformer. In Wouters and Vanden Berghe (2001), a monaural two-microphone adaptive beamforming algorithm has been developed and evaluated with LAURA CI users. For a single noise source at 90°, highly significant improvements in speech understanding, corresponding to an SNR improvement of about 10 dB, were observed. Similar results were found with normal hearing subjects and hearing aid users in Wouters, Vanden Berghe and Maj (2002).

Until now, the application of noise reduction algorithms in commercial CIs was mainly limited to the use of a hardware directional microphone. In 1997, a bilateral noise reduction system, called the Audallion BEAMformer, was marketed by Cochlear Ltd. as a preprocessing device (consisting of a digital signal processor) for the Nucleus-22 CI system (Cochlear Ltd., 1997). The Audallion BEAMformer discriminates between signals coming from the front and the back based on amplitude and phase differences between the outputs of two directional microphones, one at each ear. Sounds coming from the back are attenuated, while sounds coming from the front are passed through. In 2005, the monaural two-microphone adaptive beamformer of Wouters and colleagues (2001), referred to as BEAM, was implemented in the BTE speech processor of Cochlear's Nucleus Freedom CI system. In this paper, we present the results of a double-blind evaluation of the two-microphone adaptive beamformer BEAM and the standardly used hardware directional microphone with five adult Nucleus CI users. In contrast to the study in Wouters and colleagues (2001), the speech reception performance is not only assessed for a single noise source at 90°, but also for a more complicated, triple noise source scenario. Furthermore, the noise reduction programs have been subjectively evaluated by the subjects in real-life situations. The CI subjects were asked to compare both programs in different scenarios during a 2-wk trial period at home and to evaluate them by means of the SSQ questionnaire (Gatehouse & Noble, 2004). In addition, three of the five subjects were exposed to controlled (in the lab) and real-life noisy situations (pub, street, bus) and asked to rate the speech perception with both programs.

METHODS

The benefit of the two-microphone adaptive beamformer BEAM relative to the standard directional microphone on the speech understanding in noise by CI users was assessed based on speech understanding tests in the lab as well as subjective evaluations in real-life scenarios.

Adaptive Noise Reduction Algorithm Beam

Figure 1 depicts the two-microphone adaptive beamformer BEAM that is evaluated in this study. The beamformer combines a directional microphone (i.e., the front microphone on the BTE) and an omnidirectional microphone (i.e., the rear microphone). The distance between the directional microphone ports is 0.7 cm. The distance between the front port of the directional microphone and the omnidirectional microphone is 1.9 cm. The two-microphone beamformer, which is based on the Generalized Sidelobe Canceller of Griffiths and Jim (1982), consists of a fixed spatial preprocessor and an adaptive noise cancellation (ANC) stage.

The spatial preprocessor creates a speech reference and a noise reference. The noise reference is created by steering a zero towards the direction of the desired signal, i.e., the front. The noise reference signal is computed by filtering the omnidirectional microphone signal and subtracting it from a delayed version of the directional microphone signal. The speech reference is created by adding the filtered omnidirectional microphone signal to the delayed version of the directional microphone signal. A fixed FIR filter was used in the spatial preprocessor. The filter coefficients have been determined based on recordings of a speech-weighted noise signal at 0° with two Freedom speech processors on a Cortex MK2 artificial head in an anechoic room: the fixed filter was designed so that the energy of the noise reference signal was minimized.

The ANC attenuates the residual noise in the speech reference. The ANC is updated with the Normalized Least Mean Squares algorithm (NLMS).

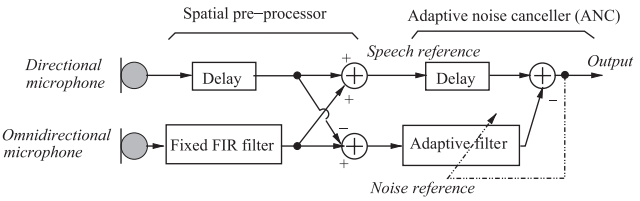


Fig. 1. Two-microphone adaptive beamformer BEAM.

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The ANC minimizes the output power while the spatial preprocessor is designed to avoid so-called speech leakage in the noise reference. However, because of microphone mismatch, head movements and room reverberation, the speech signal may leak into the noise reference, resulting in speech distortion. To limit speech distortion, the ANC is adapted during periods of noise-only (Greenberg & Zurek, 1992; Hoshuyama, Sugiyama & Hirano, 1999; Van Compernolle, 1990; Herbordt & Kellermann, 2002). The speech detection algorithm described in Vanden Berghe and Wouters (1998) is used to determine periods of noise-only.

Evaluation Procedure

Subjects • Five adult Nucleus CI users, one female and four male subjects, participated in the evaluation. It was required that the CI subjects could reach 50–60% speech understanding in quiet with the LIST sentences (van Wieringen & Wouters, 2005; van Wieringen & Wouters, Reference Note 2) to achieve an adaptive SRT in noise. Three Freedom speech processors (with serial numbers 2899, 3148, 4350) were used for the evaluation. In contrast to the study in Wouters and colleagues (2001) where the BTE was put on a mannequin to carry out the perceptual tests, the BTE was worn by the subject during the evaluation. The speech processor was programmed according to the clinical fitting data of each patient. All subjects use the ACE (Advanced Combination Encoders) speech strategy (Vandali, Whitford, Plant & Clark, 2000). The volume of the speech processor (value between 0 and 9) was adjusted to a comfortable loudness. The sensitivity was set to its default, i.e., 12.

Table 1 describes some relevant information for each subject and indicates which of the three BTEs was used by which subject. Two noise reduction programs were provided, namely, the hardware directional microphone (i.e., the directional microphone signal in Figure 1) and the two-microphone adaptive beamformer BEAM. The order of the programs was randomized over the different subjects and was only divulged to the researchers carrying out the evaluations and the CI users after completion of the study.

Set-up of the lab experiments • The speech understanding tests in the lab were performed in a standard office room. To limit the amount of reverberation, sound-absorbing panels were attached to the ceiling of the room. The average reverberation time T_{60} of the room, measured with a speech-weighted noise signal, equals 0.52 sec. The estimated intelligibility-weighted direct-to-reverberant ratio was 7 dB at 1 meter. The desired speech signal and noise signal(s) were presented through identical loudspeakers (Yamaha CBX-S3) positioned at a distance of 1 meter from the patient's head: the speech source in front of the head, the noise sources at an angle θ with respect to the speech source. The loudspeakers were calibrated separately such that they produced the same sound level for a speech-weighted noise signal at a reference point corresponding to the center of the patient's head. Two noise configurations were considered: a single noise source at 90° (on the side of the patient's CI) and three noise sources at 90°, 180° and 270°. The noise signals in the triple noise source scenario were uncorrelated and all had the same sound level at the reference point.

Presentation of the speech and noise materials was controlled from an adjacent room by means of two CD players and two audiometers. The signals, produced by two CD players, were amplified through a MADSEN OB822 audiometer (the speech signal and noise signal at 90°) and an AMPLAID 309 audiometer (the noise signals at 180° and 270°) before being sent to the loudspeakers. The maximum amplification of the audiometers corresponds to a sound pressure level of 90 dB SPL at the reference point in the test room.

Test protocol • Each test trial consisted of a pre- and post-test in the lab and a 2-wk trial period at

TABLE 1. Some details of the Nucleus CI users who participated in the evaluation, i.e., serial number of the BTE, volume setting, age in years (yr), duration of profound deafness in years (yr), etiology of hearing loss, implant experience in years (yr) and number of active electrode channels. The speech reception threshold (SRT) in quiet (dB SPL) for the directional microphone of the Nucleus Freedom CI system is also shown. The SRT was measured with the LIST sentences (van Wieringen and Wouters, 2005; van Wieringen, Wouters, Reference Note 2)

Subject	Serial number of the BTE	Volume setting	Age (yr)	Duration of profound deafness (yr)	Etiology of hearing loss	Implant experience (yr)	Number of active electrode channels	SRT in quiet (dB SPL)
A	2899	9	48	3	Unknown	9	22	51.7
B	3148	9	56	6	Work	3	22	49.9
C	4350	7/5	53	1	Progressive	1	20	49.5
D	3148	4	54	1	Cochlear otosclerosis	7	20	51.4
E	2899	6	35	<2	Unknown	10	22	50.2

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home. Between the pre- and the post-test, both noise reduction programs were compared by the CI users in different scenarios during a 2-wk trial period at home and evaluated by means of the SSQ questionnaire (Gatehouse et al., 2004).

During the pre- and the post-test in the lab, the benefit of the two-microphone beamformer regarding the ability to understand speech in noise was assessed through two procedures. In the first procedure, the SRT of the CI users was measured adaptively in quiet and in background noise for the two noise reduction programs (Plomp et al., 1979). The SRT is the sound level of the speech signal at which 50% of the presented speech sentences is correctly identified. The speech material consisted of Dutch/Flemish LIST-sentences spoken by a female speaker (van Wieringen et al., 2005; van Wieringen et al., Reference Note 2). For each condition (i.e., program and noise scenario), one list of 10 sentences was selected. Different lists were used during the pre- and the post-test. The SRT was determined using a simple up-down (1 up, 1 down) procedure with a 2-dB step size. In this procedure, the speech level was varied and the noise level was kept constant. The first sentence was repeated with increasing level until it was correctly identified by the subject. The subsequent sentences were presented only once at a 2 dB lower or higher level than the previous sentence, depending on whether the latter was identified correctly or not. A response was judged correct if and only if all the key words of the sentence were repeated correctly. To compute the SRT, the speech levels of sentences 6 to 10 and the imaginary 11th sentence were averaged. The SRT was determined for two noise levels, i.e., 55 dB SPL and 65 dB SPL, to study the influence of the noise level on the SRT for the beamformer and the directional microphone.

In the second procedure, the percentage correct phoneme scores for CVC words in quiet and in noise were measured with both noise reduction programs. The CVC words were part of the Flemish recordings of the NVA list (Wouters et al., 1994). The NVA list consists of sublists of 12 Dutch/Flemish 3-phoneme monosyllables, produced by a male speaker. For each condition, i.e., program and noise scenario, one sublist was selected. Different sublists were used during the pre-and the post-test. The response for the first word was not included in the computation of the percentage correct phoneme scores. Two SNRs were tested, i.e., −5 dB and +5 dB, at a noise level of 60 dB SPL (i.e., the average of 55 dB SPL and 65 dB SPL).

In both procedures, the performance was assessed for two different noise configurations (i.e., a single noise source and three noise sources), and two different noise materials (i.e., stationary speech-weighted noise and non-stationary multitalker babble noise). The speech-weighted noise had the same long-term spectrum as the presented speech material. For the adaptive measurement of the SRT, the speech-weighted noise on the CD by van Wieringen and colleagues (Reference Note 2) was used. For the measurement of the percentage correct phoneme scores, the steady NVA speech-weighted noise was used. The multitalker babble noise was taken from the CD *Auditory Tests (Revised)* by Auditec (Reference Note 3).

Physical Evaluation

Before the perceptual tests were performed, the noise reduction performance of the two-microphone beamformer and the hardware directional microphone for a single noise source was assessed through physical measurements with two speech processors. The physical performance of noise reduction algorithms is assessed in terms of the output intelligibility weighted SNR (Greenberg, Peterson & Zurek, 1993), defined as:

$$\text{output-SNR}_{\text{intellig}} = \sum_{i} I_i(\text{output-SNR}_i) \qquad (1)$$

where output-SNR_i is the output SNR (in dB) in the i-th one third octave band. The band importance function I_i expresses the importance of the i-th one-third octave band with center frequency f_i^c for intelligibility. The center frequencies f_i^c and the values I_i are defined in ANSI S3.5-1997 (Acoustical Society of America, Reference Note 4). The CI-specific signal processing modules such as the manual sensitivity control and the automatic gain control (AGC) were disabled during the physical measurements. Hence, only the effect of the beamformer was measured.

The measurements were performed in an anechoic room and an office room with $T_{60} = 0.76$ sec. To predict the average performance based on directivity only, stationary speech-weighted noise was used as speech and noise signal in the physical measurements. During the measurements, the speech processor was mounted on the right ear of a Cortex MK2 artificial head. Both the speech and the noise source had a level of 70 dB SPL at the center of the artificial head.

To determine the output-$\text{SNR}_{\text{intellig}}$, the speech and the noise signal were recorded separately. First, the coefficients of the beamformer were adapted to the noise signal until convergence was achieved. Then, the output speech and noise level were determined by freezing the filter coefficients of the beamformer and presenting speech-weighted noise from the speech and the noise directions, respectively.

Figure 2 depicts the output-$\text{SNR}_{\text{intellig}}$ of the directional microphone and the beamformer, averaged over the two prototype devices, as a function of

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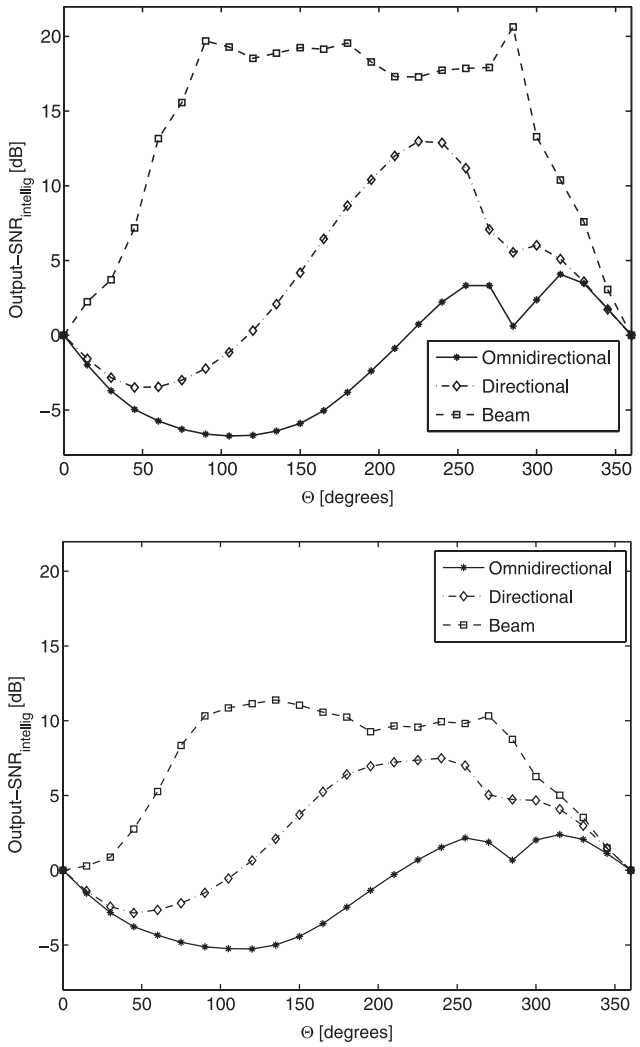


Fig. 2. Average output-SNR$_{intellig}$ of the omnidirectional (*) and directional microphone (◇) and the two-microphone beamformer BEAM (□) as a function of the angle θ of the noise source. a) Anechoic room; b) Office room with T_{60} = 0.76 sec.

the angle θ of the noise source. Figure 2(a) depicts the performance in an anechoic room; Figure 2(b) shows the performance in the office room. As a reference, the output-SNR$_{intellig}$ of the omnidirectional microphone is also depicted. For all angles θ, the beamformer reduces more noise than the directional microphone. The beamformer has its main advantage over the directional microphone for noise sources between 50° and 180°, i.e., at the side of the speech processor. Reverberation has a large effect on the performance of the beamformer. At 90°, the improvement of the beamformer relative to the directional microphone equals 12 dB in the office room and 22 dB in the anechoic room. Two factors contribute to the improvement of the beamformer compared to the directional microphone. First, the beamformer is adaptive and second, the beamformer uses an additional omnidirectional microphone. For a single noise source in a room with low to medium reverberation, the contribution of the additional microphone is limited compared to the improvement due to adaptivity. In the other extreme case of diffuse noise, the additional omnidirectional microphone results in a slight improvement (around 1 dB) of the beamformer compared to the directional microphone (Maj, Wouters & Moonen, 2004). The noise reduction performance with the three Freedom speech processors used in the perceptual evaluation may be slightly worse because of differences in the mechanics of the microphone housings as well as microphone mismatch (compare with section *Adaptive Noise Reduction Algorithm*).

Perceptual Evaluation: Results and Discussion

This section discusses the results of the speech understanding tests carried out in the lab, i.e., the measurement of the SRTs and the percentage correct phoneme scores with both noise reduction programs in quiet and in noise. The scores of the pre- and the post-test have been averaged because no statistically significant difference was found between the two test sessions (see section *Reliability*).

SRT in quiet and in noise • Table 2 and Table 3 show the results of the adaptive SRT measurements with the LIST sentences. The SRTs in quiet and in noise of the five patients as well as the average SRT and standard deviation are depicted for the two noise reduction programs. In addition, the differences between the two programs are shown. The results in quiet are expressed as actual SRT values in dB SPL; the results in noise are expressed as SNRs in dB and can be found in Table 2 for the single noise scenario and in Table 3 for the triple noise scenario. Results for the different noise levels (55 dB SPL and 65 dB SPL) and noise materials (speech-weighted noise and babble noise) are displayed separately.

The SRT values in Table 2 and Table 3 that are denoted with stars are not accurate. To reach the SNR of 50% correct identification with the directional microphone in babble noise at 65 dB SPL, high speech levels were required. During the measurement of some SRT values (denoted with *), the speech level could not be increased until this 50% identification point was reached because of practical limitations of the amplification system. In addition, at these high presentation levels, the AGC began to compress the speech (the presentation level equivalent to the AGC threshold is determined by the manual sensitivity setting). As a result, the true

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TABLE 2. Results of the adaptive SRT measurements of sentences in quiet (in dB SPL), speech-weighted noise and multitalker babble noise (SNR in dB) for a single noise source at 90° with the directional microphone (Dir. Mic) and the beamformer BEAM

		SRT in noise, as SNR (dB) Dir. Mic/BEAM/**Improvement**			
		Speech-weighted noise		Babble noise	
Subject	SRT in quiet (dB SPL) Dir. Mic/BEAM/**Improvement**	55 dB SPL	65 dB SPL	55 dB SPL	65 dB SPL
A	51.7/55.5/**−3.8**	4.4/−2.7/**7.0**	12.0/−6.4/**18.4**	15.4/5.4/**10.0**	22.2*/8.0/**14.2***
B	49.9/51.0/**−1.1**	−1.9/−3.5/**1.7**	−1.5/−6.2/**4.7**	9.0/1.0/**8.0**	12.5/−2.7/**15.2**
C	49.5/50.7/**−1.2**	7.0/−3.0/**10.0**	3.2/−8.2/**11.4**	10.0/12.9/**−2.9**	22.2*/8.2/**14.0***
D	51.4/48.7/**2.7**	7.0/−4.4/**11.4**	11.7/−2.7/**14.3**	15.7/2.7/**13.0**	22.5*/7.0/**15.5***
E	50.2/50.9/**−0.7**	6.9/0.9/**6.0**	15.2*/−2.5/**17.7**	16.2/8.2/**8.0**	22.9*/2.5/**20.4***
Mean	50.5/51.4/**−0.8**	4.7/−2.5/**7.2**	8.2/−5.2/**13.4**	13.2/6.0/**7.2**	20.5*/4.6/**15.9***
SD	1.0/2.5/**2.3**	3.8/2.0/**3.8**	7.1/2.5/**5.7**	3.4/4.7/**6.0**	4.5/4.7/**2.6**

** Exact value could not be determined: audiometer was out of range and the speech sound pressure level was entering the compression range of the AGC.*

SRT for the conditions marked with a star can be somewhat different.

Reliability • The reliability of the results was examined by comparing pre- and post-test-results. A repeated measures analysis of variance (ANOVA), including all results, revealed that results of the two test moments did not differ significantly ($F = 0.358$, $p = 0.582$). Also a high Pearson correlation was found between the two measurements ($r = 0.922$, $p < 0.001$).

SRT in quiet • The mean SRT in quiet corresponded to 51.4 dB SPL and 50.5 dB SPL for the beamformer and the hardware directional microphone, respectively. The difference between the two programs was insignificant (ANOVA, $F = 0.625$, $p = 0.473$). Theoretically, one does also not expect a difference between the programs, as the beamformer is not intended to affect speech performance in quiet.

Speech understanding in noise • For all subjects, the SRT in noise was generally improved by the beamformer.* The average benefit in SRT obtained with the two-microphone beamformer BEAM compared to the hardware directional microphone is visualized in Figure 3. The left part of the graph shows the results for the noise level of 55 dB SPL, while the results for 65 dB SPL can be found at the right-hand side of the figure. Each bar represents one condition (one noise material in one noise scenario). Error bars depict the standard deviation. As can be seen in Figure 3, the beamformer improved the SRT in noise on average by 2 dB to 17 dB compared to the hardware directional microphone. A benefit was observed in all noise conditions, with the largest one in the condition with a single noise source at 90° and a level of 65 dB SPL (average improvement of 13.4 dB for speech-weighted noise and 15.9 dB for babble noise). A repeated measures analysis of variance (ANOVA) proved the difference between two noise reduction algorithms to be significant ($F = 130.141$, $p < 0.001$).

Influence of the noise scenario on speech intelligibility • In addition to the microphone type, speech understanding in general (i.e., the absolute SRT with the directional microphone and the beamformer) seemed to depend on the type of noise. Babble noise is known to be more disturbing than stationary noise and therefore the SRTs were significantly poorer for babble noise than for stationary noise ($F = 264.275$, $p < 0.001$). The level of the noise did not have a significant influence on the speech reception thresholds, expressed as signal-to-noise ratios in dB ($F = 1.011$, $p = 0.371$). Although one could expect the three noise sources at 90°/180°/270° to be less disturbing than the single noise source at 90° because of head shadow, speech understanding did not seem to depend on the noise configuration ($F = 0.006$, $p = 0.942$).

The difference in speech understanding between the beamformer and the hardware directional microphone was significantly smaller in speech-weighted noise than in babble noise ($F = 12.121$, $p = 0.025$). For speech-weighted noise, the average improvement equaled 1.5 dB to 13.5 dB; for multitalker babble noise, the average improvement equaled 5.3 dB to 15.9 dB (compare with Fig. 3). This might be due to the larger disturbing effect of babble noise on speech understanding in general.

The improvement in SRT by the beamformer was larger for 65 dB SPL than for 55 dB SPL. For 55 dB SPL, an average improvement between 1.5 dB and 7.2 dB was obtained, whereas for 65 dB SPL, the average improvement lay between 6.5 dB and 15.9 dB. The SRTs for a noise level of 55 dB SPL were quite close to the SRTs in quiet (i.e., the optimal

*There are two exceptions, i.e., the scenario with three speech-weighted noise sources at 55 dB SPL for subject A and the single babble noise source at 55 dB SPL for subject C.

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TABLE 3. Results of the adaptive SRT measurements of sentences in speech-weighted noise and multitalker babble noise (SNR in dB) for 3 noise sources at 90°, 180°, and 270° with the directional microphone (Dir. Mic) and the beamformer BEAM

	SRT in noise, as SNR (dB) Dir. Mic/BEAM/**Improvement**			
	Speech-weighted noise		Babble noise	
Subject	55 dB SPL	65 dB SPL	55 dB SPL	65 dB SPL
A	−1.4/4.3/**−5.7**	3.0/−6.0/**9.0**	10.0/3.7/**6.4**	19.9*/6.4/**13.5***
B	6.5/2.5/**4.0**	1.5/−4.9/**6.4**	14.8/10.3/**4.5**	11.9/3.5/**8.4**
C	2.9/0.0/**2.9**	2.0/−2.7/**4.7**	10.7/6.2/**4.5**	18.5*/5.8/**12.7***
D	4.0/0.6/**3.4**	8.0/0.7/**7.4**	10.7/3.0/**7.7**	22.5*/9.4/**13.2***
E	2.9/0.2/**2.7**	2.9/−2.5/**5.4**	11.2/7.9/**3.3**	21.5*/10.5/**10.0***
Mean	3.0/1.5/**1.5**	3.5/−3.1/**6.5**	11.5/6.2/**5.3**	18.9/7.1/**11.6***
SD	2.8/1.8/**4.0**	2.6/2.6/**1.7**	1.9/3.0/**1.7**	4.2/2.8/**2.3**

** Exact value could not be determined: audiometer was out of range and the speech sound pressure level was entering the compression range of the AGC.*

speech intelligibility for the CI patients). This leaves little room for improvement by the beamformer and explains why the benefit achieved at this noise level was significantly smaller than the benefit at a noise level of 65 dB SPL ($F = 33.188$, $p = 0.005$).

While the noise configuration did not have a significant influence on the absolute speech reception thresholds, the beamformer benefit was significantly larger for a single noise source at 90° than for multiple noise sources at 90°/180°/270° ($F = 14.810$, $p = 0.018$). For a single noise source at 90°, average improvements in SRT between 7.2 dB and 15.9 dB were obtained with the beamformer, while for three noise sources, the average improvement lay between 1.5 dB and 11.6 dB. This can be explained by the fact that because of head shadow, the total noise level reaching the microphones was lower in the three-noise scenario than in the single-noise scenario. Therefore the maximum improvement that could be obtained with a better noise reduction algorithm was reduced (compare with previous paragraph). Moreover, the two-microphone beamformer can better suppress a single noise source around 90° than multiple noise sources.

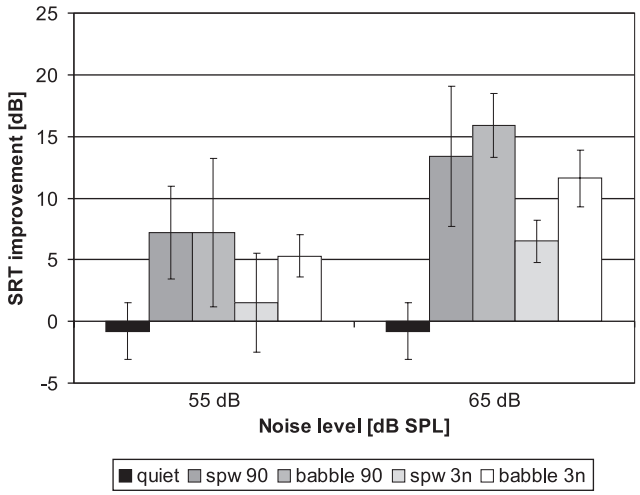


Fig. 3. Average benefit in SRT [dB] of the two-microphone beamformer BEAM compared to the hardware directional microphone in quiet and in the different noise scenarios, i.e., speech-weighted noise at 90°(spw 90), babble noise at 90° (babble 90), speech-weighted noise at 90°, 180°, 270° (spw 3n) and babble noise at 90°, 180°, 270° (babble 3n).

Percentage Correct Phoneme Scores for CVC Words

Individual percentage correct phoneme scores, as well as the average scores and standard deviation, for the fixed level CVC-test can be found in Table 4 and Table 5 for the single noise source and the triple noise sources scenario, respectively. Pre- and post-test results were averaged. Results are shown for speech at 55 dB SPL and 65 dB SPL in quiet and in noise with a fixed level of 60 dB SPL (i.e., SNR of −5 dB and +5 dB).

Reliability • Post-test scores did not differ significantly from pre-test scores, as is proven by the repeated measures ANOVA ($F = 0.04$, $p = 0.856$). The Pearson correlation coefficient also indicated a significant concordance ($r = 0.688$, $p < 0.001$).

Speech understanding in quiet • With a mean percentage correct score of 71% for the beamformer and 72% for the directional microphone, the noise reduction processing again did not lead to a significant difference in speech understanding in quiet ($F = 0.002$, $p = 0.962$). This outcome follows expectations.

Speech understanding in noise • For almost all subjects and conditions an improvement by the beamformer was found.* Figure 4 depicts the average benefit (±one standard deviation) for all subjects, of the two-microphone beamformer compared

*For subject C and D, a reverse effect was found for the three babble noise sources and the speech level at 65 dB SPL.

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TABLE 4. Percentage correct phoneme scores for CVC words in quiet and for a single noise source at 90° with the directional microphone (Dir. Mic) and the beamformer BEAM

	Score in quiet (%) Dir. Mic/BEAM/**Improvement**		Scores in noise (%) Dir. Mic/BEAM/**Improvement**			
			Speech-weighted noise		Babble noise	
Subject	55 dB SPL	65 dB SPL	55 dB SPL	65 dB SPL	55 dB SPL	65 dB SPL
A	89.5/83.5/−**6.0**	85.0/73.0/−**12.0**	15.0/82.0/**67.0**	50.0/56.0/**6.0**	1.5/24.0/**22.5**	38.0/47.0/**9.0**
B	71.5/68.5/−**3.0**	71.5/76.0/**4.5**	41.0/58.0/**17.0**	70.0/76.0/**6.0**	15.0/33.0/**18.0**	48.5/65.5/**17.0**
C	64.0/65.5/**1.5**	73.0/70.0/−**3.0**	24.0/62.5/**38.5**	45.5/62.5/**17.0**	0.0/47.0/**47.0**	33.0/70.0/**37.0**
D	67.0/62.5/−**4.5**	71.5/76.0/**4.5**	24.0/48.5/**24.5**	40.5/67.0/**26.5**	3.0/37.5/**33.5**	25.5/71.5/**46.0**
E	67.0/74.5/**7.5**	68.5/71.5/**3.0**	7.5/65.5/**58.0**	37.5/68.5/**31.0**	1.5/24.0/**22.5**	25.5/57.5/**32.0**
Mean	71.8/70.9/−**0.9**	73.9/73.3/−**0.6**	22.3/63.3/**41.0**	48.7/66.0/**17.3**	4.2/33.1/**28.9**	34.1/62.3/**28.2**
SD	10.3/8.3/**5.5**	6.4/2.7/**7.1**	12.5/12.3/**21.3**	12.8/7.4/**11.5**	6.1/9.7/**11.8**	9.6/10.1/**15.0**

to the hardware directional microphone, in percentage correct phoneme scores in noise, for the different SNRs (–5 dB or +5 dB), noise materials (speech-weighted noise or babble noise) and configurations (noise at 90° or at 90°/180°/270°). The beamformer resulted in an average improvement in percentage correct phoneme scores between 3.1% and 41% compared to the hardware directional microphone. ANOVA revealed a highly significant difference between the hardware directional microphone and the beamformer ($F = 192.243$, $p < 0.001$).

Influence of the noise scenario on speech intelligibility • As could be expected, speech understanding results were better when the SNR was high. This is confirmed by the ANOVA ($F = 82.195$, $p = 0.001$). Furthermore, the same observations were made as in the adaptive SRT measurements: babble noise led to significantly worse speech intelligibility ($F = 202.123$, $p < 0.001$) but a different noise configuration (90° or 90°/180°/270°) did not give rise to significantly different percentages ($F = 0.75$, $p = 0.435$).

The improvement by the two-microphone beamformer was particularly large for an SNR of −5 dB. For this SNR, the beamformer gave rise to an average improvement in percentage correct phoneme scores of 17.1% to 41% compared to the directional microphone. For an SNR of +5 dB, the improvement was smaller but still significant (e.g., up to 29% for the babble noise source at 90°). Figure 5 demonstrates that the percentage correct phoneme scores for the directional microphone at an SNR of +5 dB were close to the results in quiet (71%–72%). As a result, there was little room for improvement by the beamformer at this SNR. The effect of the noise level (or SNR) on the improvement by the beamformer was also shown by ANOVA ($F = 9.012$, $p = 0.04$).

Other interaction effects were less strong than observed in the adaptive testing, or even absent: babble noise did not lead to a greater beamformer benefit in the percentage correct phoneme scores ($F = 0.04$, $p = 0.851$) and the effect of the noise configuration on the beamformer benefit was too small to be significant ($F = 7.316$, $p = 0.054$).

Self-report Evaluations: Results and Discussion

Between the pre- and post-test in the lab, the noise reduction algorithms were evaluated by the CI

TABLE 5. Percentage correct phoneme scores for CVC words. Three noise sources at 90°, 180°, and 270° with the directional microphone (Dir. Mic) and the beamformer BEAM

	Scores in noise (%) Dir. Mic/BEAM/**Improvement**			
	Speech-weighted noise		Babble noise	
Subject	55 dB SPL	65 dB SPL	55 dB SPL	65 dB SPL
A	30.0/59.5/**29.5**	59.0/76.0/**17.0**	16.5/44.0/**27.5**	53.5/54.0/**0.5**
B	24.0/42.5/**18.5**	62.5/70.0/**7.5**	0.0/22.5/**22.5**	48.5/67.0/**18.5**
C	43.5/45.5/**2.0**	58.0/65.6/**7.5**	16.5/33.0/**16.5**	53.0/45.5/−**7.5**
D	30.0/53.3/**23.5**	54.5/68.5/**14.0**	16.5/36.5/**20.0**	48.0/43.5/−**4.5**
E	36.5/48.5/**12.0**	56.0/60.5/**4.5**	6.0/30.0/**24.0**	46.5/55.0/**8.5**
Mean	32.8/49.9/**17.1**	58.0/68.1/**10.1**	11.1/33.2/**22.1**	49.9/53.0/**3.1**
SD	7.4/6.7/**10.6**	3.1/5.7/**5.2**	7.7/7.9/**4.1**	3.2/9.3/**10.5**

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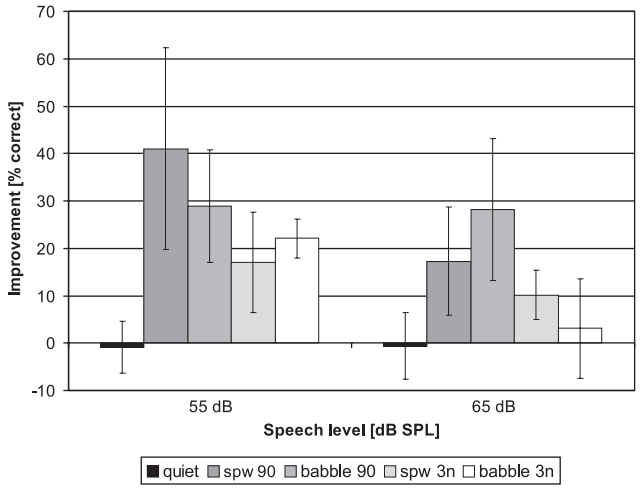


Fig. 4. Average benefit in percentage correct phoneme scores of the beamformer relative to the directional microphone in quiet and in the different noise scenarios, i.e., speech-weighted noise at 90° (spw 90), babble noise at 90° (babble 90), speech-weighted noise at 90°, 180°, 270° (spw 3n) and babble noise at 90°, 180°, 270° (babble 3n).

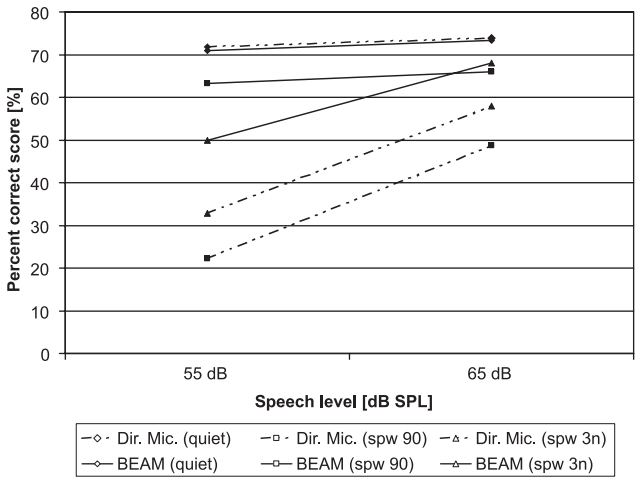


Fig. 5. Absolute percentage correct phoneme scores in quiet (◇) and in speech-weighted noise (one noise source at 90° (spw 90; □); three noise sources at 90°, 180° and 270° (spw 3n; △)) with the directional microphone (dashed-dotted line) and with the beamformer BEAM (solid line).

subjects through an SSQ questionnaire (Gatehouse et al., 2004). Only the Speech and Qualities parts of the SSQ questionnaire were used (14 and 18 questions,* respectively). In addition, three of the five subjects were exposed to both controlled and real-life noisy listening scenarios and were asked to compare speech understanding with both noise reduction programs.

SSQ • Below, we summarize the results of the SSQ questionnaire. The criterion used to establish a preference for one program over the other is that the program is preferred by at least two subjects and the mean difference in SSQ units with the other program is at least 0.4 units.

The majority of the SSQ Speech and Qualities questions (23 out of the 32 items) did not reveal a preference for one noise reduction program over the other (i.e., items 2–4, 7–9, 13–14 of the Speech part and items 1–5, 7–14, 18–19 of the Qualities part). Except for item 4 of the Speech part ("following a conversation in a busy restaurant while you can see every person of the group"), none of these items are related to the ability to understand speech in noise. They focus on speech understanding in quiet, following a conversation on the telephone, the naturalness of sounds and the ability to distinguish between different sounds.

The other 9 questions of the SSQ questionnaire (i.e., questions 1, 5, 6, 10, 11 and 12 of the Speech part and questions 6, 16 and 17 of the Qualities part) showed a preference for the beamformer by 2 to 4 subjects. For one of the subjects no preference was found. The mean difference between the noise reduction programs for these questions was between 0.4 and 0.7 units. Almost all of the questions where the subjects showed a preference for the beamformer are related to the ability to understand speech in noise (i.e., understanding speech in the presence of interfering speakers, car noise or a television). Exceptions are question 12 of the Speech part ("following a conversation without missing the start of what each new speaker is saying") and question 6 of the Quality part ("ability to distinguish between different sounds such as a car and a bus").

None of the questions demonstrated a preference for the directional microphone.

The results from the SSQ questionnaire seem to be less convincing than those from the speech understanding tests in the lab. This is partly due to the fact that not all questions are adequate in revealing a difference between noise reduction programs: only a few questions focus on the ability to understand speech in noise. Besides that, the listening conditions that subjects encountered in daily life during the 2-wk trial period at home were often less noisy than the ones created in the lab. Some of the subjects even spent most of the time in quiet surroundings. This may have led to an underestimation of the beamformer's effect by the questionnaire.

Subjective testing in the lab and in Real-life Situations • To obtain subjective results in a more controlled noisy environment, three subjects (A, B and C) were asked to subjectively evaluate both programs in the lab test setup. Among them was the subject who did not show any preference for the

*Item 15 of the Qualities part is only applicable to bilateral hearing aid or cochlear implant users. Hence, it was not used in this evaluation.

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beamformer in the SSQ questionnaire. The speech material consisted of the CVC words. The subjects were exposed to all different noise conditions (speech-weighted noise at 90°, speech-weighted noise at 90°/180°/270°, babble noise at 90° and babble noise at 90°/180°/270°) in a randomized order. The noise level was set to 65 dB SPL. The speech level was adjusted by the subjects to a level where they could discriminate the speech with a reasonable intelligibility. Subjects switched back and forth between the two programs and rated on a 0–10 scale—the speech understanding, going from no understanding (0) to perfect understanding (10).

For all noise scenarios, all three subjects preferred the beamformer over the directional microphone. The average scores on a 0–10 scale for the hardware directional microphone and for the beamformer, as well as the difference in the average scores of both programs are shown in Table 6. The average benefit of the beamformer for speech understanding ranged from 2 to 3.6 units for the different conditions. The benefit was larger for the single noise source than for the multiple noise sources, which is in agreement with the results of the adaptive SRT measurements. The difference between the beamformer and hardware directional microphone is more pronounced than the differences found in the SSQ questionnaire. A reason for this could be that, within the 2 wk the subjects evaluated the noise reduction programs at home, subjects did not spend much time in noisy situations where the beamformer should lead to an advantage in speech understanding.

To assess this possibility, three subjects (A, B and C) were taken out in real-life situations and again asked to rate their perceived speech understanding in the vicinity of a speaker for both programs on a 0–10 scale. Four situations were selected for this purpose: a ride by bus, a bus station, a shopping street and a pub/restaurant. Table 7 shows the results of this evaluation. The mean benefit in rated speech understanding obtained with the beamformer ranged from 0.25 to 1.17 out of a possible score of ten. The most important conclusion to be drawn from this test session is that in real-life noisy conditions the subjective benefit from the beamformer is smaller than the benefit in controlled laboratory conditions. Possible explanations are that the real-life listening conditions were less noisy and/or more reverberant than the laboratory conditions and that the CI subjects used lip-reading.* Note that the absolute scores for speech understanding with the directional microphone were lower in the lab (on average 3 out of 10) than in real-life (on average 7 out of 10). Hence, there was less room for improvement by the beamformer in the real-life test conditions.

TABLE 6. Results (scores out of 10 for speech understanding) of the subjective evaluation of the hardware directional microphone (Dir. Mic) and the beamformer BEAM in controlled noise scenarios in the lab. Noise level of 65 dB SPL

	Scores for speech understanding: Dir. Mic/BEAM/**Improvement**			
	Speech weighted		Babble	
	90°	3 noises	90°	3 noises
Mean	2.7/6.4/**3.7**	3.9/6.2/**2.3**	3.0/6.0/**3.0**	4.0/6.0/**2.0**
SD	0.8/1.3/**0.8**	0.8/0.5/**0.9**	1.8/1.3/**0.5**	1.3/1.3/**0.0**

TABLE 7. Results (scores out of 10 for speech understanding) of the subjective evaluation of the directional microphone (Dir. Mic) and the beamformer BEAM in real-life noise scenarios

	Dir. Mic/BEAM/**Improvement**			
	Bus	Bus station	Street	Pub/restaurant
Mean	6.1/6.8/**0.7**	5.9/7.1/**1.2**	5.3/6.0/**0.7**	7.0/7.3/**0.3**
SD	0.4/0.3/**0.7**	0.4/0.1/**0.3**	0.4/0.3/**0.3**	1.1/0.5/**0.7**

The SSQ and subjective evaluations suggest that CI users may prefer the beamformer in real-life noisy listening conditions. However, the effect of the beamformer for the subjective evaluations is smaller than for the speech understanding tests in the lab. To show statistically significant differences between the beamformer and the directional data on the subjective data, a larger sample size is needed. For this reason, no statistical tests were performed on the SSQ and the subjective data. Given the small group of subjects in this subjective evaluation, the preference for the beamformer may not be generalized to all CI users.

CONCLUSIONS

In this study, the two-microphone adaptive beamformer BEAM in the new Nucleus Freedom CI system has been evaluated through a double-blind evaluation with five CI users. This is the first implementation of a monolateral (adaptive) noise reduction strategy in a mainstream commercial CI. Speech understanding tests in the lab demonstrated significant improvements of both the SRT in noise (average improvement of 5–16 dB) and the percentage correct phoneme scores (average improvement of 10–41%) with BEAM compared to the standard hardware directional microphone in all tested conditions. The largest benefit was obtained for the

*The experiments in real-life were performed during the day between 9.00 and 16.00. The listening conditions were not always found to be that noisy.

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noise level of 65 dB SPL and the single noise source at 90°. Self-report evaluations in controlled and real-life scenarios suggested a possible preference for the beamformer in noisy environments.

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