EFFECTS OF "DIRECTIONAL" SIGNAL PROCESSING IN HEARING INSTRUMENTS ON SPEECH UNDERSTANDING IN NOISE AND LOCALIZATION

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ABSTRACTS

Results of two studies will be reported at the conference. Firstly, the enhancement of speech understanding in noisy conditions will be discussed for multi-microphone configurations combined with adequate signal processing in behind-the-ear (BTE) applications. Secondly, adverse effects of bilateral hearing aids on directional hearing in the frontal horizontal plane will be illustrated.

Using 2 microphones and additional adaptive signal processing with limited complexity, it has been shown that speech understanding in different types of jammer noises (multiple talkers) and sound scenes (sounds simultaneously from different directions) can be improved to some extent. In moderately reverberant rooms improvements of about 10dB can be obtained for a single jammer source, relative to a single directional microphone. Some results obtained with adaptive noise reduction strategies will be presented:

i) adaptive beamforming (or generalized sidelobe cancellers) which provide good results, for a number of listening situations significantly better than the adaptive zooms implemented in some hearing aids, and using low-complexity algorithms. This technique has recently been introduced in a commercial cochlear implant system.
ii) adaptive optimal filtering techniques, which, at the expense of a slightly higher complexity, provide an increased robustness against changes in speaker positions and microphone characteristics.

The localization abilities in the horizontal plane have been evaluated with bilateral hearing aid users using different stimuli and different noise scenarios. The main research questions raised were (i) how do bilateral hearing aid users perform on a localization task, relative to normal hearing subjects, (ii) do bilateral hearing aids preserve localization cues, (iii) is there an influence of state of the art noise reduction algorithms, more in particular an adaptive directional microphone configuration, on localization performance? The hearing aid users were tested without and with their hearing aids, using both a standard omnidirectional microphone and an adaptive directional microphone configuration. Results of this research will be discussed.

1. Introduction

In the last decade microphone directionality has been widely exploited for use in commercial hearing aids. This evolution has been one of the changes in hearing aids through the last decade that a lot of hearing impaired users have convincingly reported to be of importance in daily life. Although the implementations in hearing instruments are still rather simple, the enhanced possibilities of digital signal processing in hearing instruments may open new avenues for enhancement of speech understanding in noisy environments.

As illustrated in figure 1, listening environments can be rather complex. Besides the desired direct speech sound of a talker, also jammer noises or other sources, as well as the reflected/reverberated sounds of these sources are picked up by the microphones in the hearing instrument. In general, the desired source cannot perfectly be singled out by attenuating the other sounds. Nevertheless, the combination of multi-microphone systems and advanced signal processing can at least improve speech understanding in some adverse acoustical conditions. Results of two studies that address, with a critical view, some of the important issues of hearing instrument users in daily listening environments are reported in this conference proceedings article. Parts of this article are reported in a more extensive format (Van den Bogaert et al., 2005; Spriet et al., 2005).

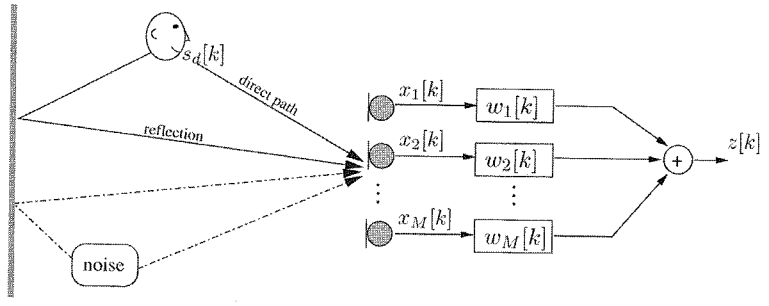

Figure 1
Sound paths in a typical listening environment of a hearing instrument user

Firstly, the enhancement of speech understanding in noisy conditions is discussed for multi-microphone configurations combined with adequate signal processing in behind-the-ear (BTE) applications. Using 2 microphone inputs and adaptive signal processing with limited complexity, it has been shown that speech understanding in different types of jammer noises (multiple talkers) and sound scenes (simultaneous sounds from different directions) can be improved to some degree. Some results obtained with an adaptive noise reduction application in a recently released cochlear implant system are described, with emphasis on the relation between laboratory results and self-report evaluations (Spriet et al., 2005).

Secondly, adverse effects of bilateral hearing aids on the directional hearing abilities of hearing aid users in the frontal horizontal plane will be illustrated. The main mechanisms used for sound localization are well known. Localization involves binaural processing

of very small differences in time (10-700s), intensity (0-20dB) and spectrum between the two ears. Extensive psychoacoustical research has been done on localization. Although a lot of localization studies have been carried out with normal hearing and hearing impaired subjects using different stimuli, very little data are available about localization with bilateral hearing aids and the binaural potential of the hearing aid user, except for a few studies (Hausler et al, 1983; Byrne et al, 1998; Noble and Byrne, 1990; Noble et al, 1994). Moreover, the effect of recently implemented digital signal processing in modern hearing aids on localization has not been thoroughly investigated yet. In recent years, adaptive signal processing is more and more being used in digital hearing aids. These techniques adapt according to changes in noise scenario or acoustic condition, but are typically designed and evaluated monaurally. An important question is whether these techniques can have an impact on binaural processes, in particular on localization. In this article a study on localization performance of bilateral BTE hearing aid users will be reported (Van den Bogaert et al., 2005).

2. Adaptive beamformer for cochlear implants

Except for the use of a hardware directional microphone, the application of noise reduction algorithms in commercial cochlear implants (CI) was until now non-existing. Cochlear Ltd. recently incorporated the two-microphone adaptive beamformer of Wouters and Vanden Berghe (2001), referred to as BEAM, in the BTE speech processor of their new Nucleus Freedom CI system. We present results of a double-blind evaluation of this adaptive beamformer relative to the standard hardware directional microphone.

2.1 Methods

For the signal processing concept of the BEAM-algorithm we refer to other publications (Wouters and Vanden Berghe, 2001; Wouters et al, 2002). This approach, here based on inputs of an omnidirectional microphone and a hardware directional microphone, has been implemented in the Freedom cochlear implant system (Spriet et al., 2005);

Subjects
Five adult Nucleus CI users, one female and four male subjects, participated in the evaluation. Freedom BTE speech processors were 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 speech strategy. 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.

Set-up
The speech understanding tests in the lab were performed in a standard office room, with T60 of 0.54s. The desired speech signal and noise signal(s) have been presented through identical loudspeakers (Yamaha CBX-S3) positioned at a distance of 1 meter from the patient's head. 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 have been considered: a single noise source at 90° (at the side of the patient's CI) and three uncorrelated noise sources at 90°, 180° and 270°.

Test protocol
Two noise reduction programs were provided, namely, the hardware directional microphone and the two-microphone adaptive beamformer BEAM. The order of the programs was randomized over the different subjects and only announced to the researchers carrying out the evaluations and the CI users after completion of the complete test procedure (double-blind).

The test procedure consisted of a pre- and post-test in the lab and a two-weeks field test at home. Between the pre- and the post-test, both noise reduction programs were compared by the CI users in different scenarios at home and evaluated by means of the SSQ questionnaire (Gatehouse and Noble, 2004). Only the Speech and Quality part of the SSQ has been used (14 and 19 questions, respectively). At the pre- and the post-test in the lab, both programs were assessed for speech understanding in in quiet and in noise 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 programs. The speech material consisted of Dutch/ Flemish LIST-sentences. The speech level was varied and the noise level was kept constant. To compute the SRT, the speech levels of the last 6 responses of a list of 10 sentences have been averaged. The SRT was determined for two noise levels, i.e., 55 dB SPL and 65 dB SPL. In the second procedure, the percentage correct phoneme scores for CVC words in quiet and in noise were measured. The CVC words are the Flemish recordings of the NVA list. Two SNRs have been tested, i.e., -5 dB and + 5dB, at a noise level of 60 dB SPL. In both procedures, the performance has been assessed for two different noise configurations and two different noise materials (i.e., stationary speech weighted noise and non-stationary multi-talker babble noise).

2.2 Results of perceptual evaluation

A repeated measures analysis of variance (ANOVA), including all results, revealed that results of the pre- and post-test moments are identical ($p = 0.509$) and a high Pearson correlation was found between the two measurements ($r = 0.922$, $p < 0.001$).

Speech understanding in noise

For all subjects, the SRT in noise is generally improved by the beamformer. The beamformer improves the SRT in noise on average by 2 to 17 dB compared to the hardware directional microphone. A benefit is observed in all noise conditions, but largest for 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 programs to be highly significant ($p = 0.001$).

Speech understanding in general (i.e., the absolute SRT) depends on the type of noise. Babble noise is more disturbing than stationary noise and leads to significantly worse results ($p < 0.001$). The difference in speech understanding between the beamformer and the hardware directional microphone is significantly smaller in speech weighted noise than in babble noise ($p = 0.014$). For speech weighted noise, the average improvement equals 1.5 to 13.5 dB; for multi-talker babble noise, the average improvement equals 5.3 to 15.9 dB. This might be due to the larger disturbing effect of babble noise on speech understanding in general.

The improvement in SRT by the beamformer is larger for 65 dB SPL than for 55 dB SPL. For 55 dB SPL, an average improvement between 1.5 and 7.2 is obtained, whereas for 65 dB SPL, the average improvement lies between 6.5 dB and 15.9 dB. The SRTs for a noise level of 55 dB SPL are closer to the SRTs in quiet (i.e., the optimal speech intelligibility by the CI patients). This leaves less room for improvement by the beamformer and explains why the benefit achieved at this noise level is significantly smaller than the benefit at a noise level of 65 dB SPL ($p = 0.002$).

While the noise configuration does not have a significant influence on the absolute speech reception thresholds, the beamformer benefit is significantly larger for a single noise source at 90° than for multiple noise sources at 90°/180°/270° ($p = 0.011$). For a single noise source at 90° average SRT improvements between 7.2 and 15.9 dB were obtained with the beamformer, while for three noise sources, the average improvement lies 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 is lower in the three-noise scenario than in the single-noise scenario.

Percentage correct phoneme scores for CVC words

Average percentage correct phoneme scores for the fixed level CVC-test are shown in figure 2 for the single noise source and the three noise sources scenario, respectively. Pre- and post-test results have been averaged. Results are shown for speech at 55 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). With a mean percentage correct score of 71 % for the beamformer and 72 % for the directional microphone, the noise reduction processing again does not lead to a significant difference in speech understanding in quiet ($p = 0.962$), as expected.

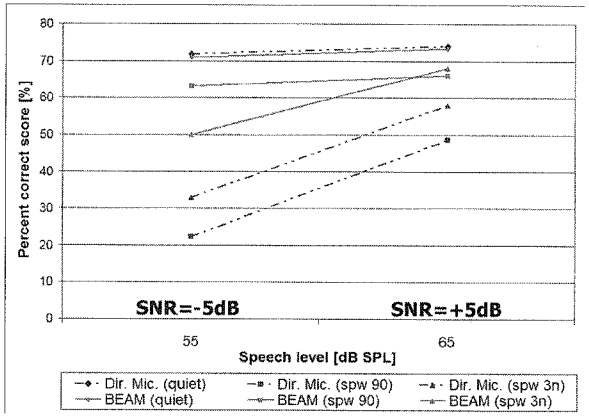

Figure 2

Absolute percentage correct phoneme scores in quiet and in speech weighted noise of 60 dB SPL (one noise source at 90° or three uncorrelated noise sources at 90°, 180° and 270°) with the directional microphone (dashed-dotted line) and with the beamformer BEAM (solid line).

In noise the two-microphone beamformer revealed a highly significant difference to the hardware directional microphone ($p < 0.001$). The beamformer results in an average improvement in percentage correct phoneme scores between 3% and 41% compared to the hardware directional microphone. Different parameters (such as the noise level, the noise material) affect the percentage correct phoneme scores and the benefit. As expected, speech understanding results are better when the SNR is high. The improvement by the beamformer is particularly large for an SNR of -5 dB. At low SNR, the beamformer gives rise to an average improvement in percentage correct phoneme scores of 17% to 41% compared to the directional microphone. For an SNR of +5 dB, the improvement is smaller but still significant (e.g., up to 29 % for the babble noise source at 90°). The improvement at the higher signal-to-noise ratio is smaller because scores are close to the results in quiet (71%-72%) and leaves little room for improvement.

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2.3 Results of self-report evaluations

Between the pre- and post-test in the lab, both programs have been evaluated by the 5 CI subjects using SSQ questionnaires. Hereafter, three of the five subjects (the 2 subjects that evaluated the beamformer most positively were not asked) have been exposed to both controlled and real-life noisy listening scenarios and were asked to compare the speech understanding with both programs.

SSQ

The questions 1, 4, 5, 6, 10, 11, 12 of the Speech part and 16, 17 of the Qualities part of the SSQ are related to the ability to understand speech in noise. For the specific content of these questions we refer to Gatehouse and Noble (2004). Although the mean difference between the two programs may seem quite small (between 0 and 0.7 units), the beamformer's benefit observed in the questionnaire is an important result. In general, the results are in favour of the beamformer, especially for questions related to speech understanding in noise. The qualities part demonstrates that the sound quality does not seem to be affected by the beamformer.

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 two-weeks field test at home are often not as noisy as the ones created in the lab. Some of the subjects even spent most of the time in quiet surroundings. This may lead to an underestimation of the beamformer's effect.

Subjective testing in the lab

In order to obtain subjective results in a more controlled noisy environment, three subjects were asked to evaluate both programs in the lab test setup. The speech material consisted of the CVC words. The subjects were exposed to all different noise conditions (speech weighted noise and babble at 90° and 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 randomized programs and rated, on an SSQ visual analog scale, the speech understanding, going from no understanding (0) to perfect understanding (10), and the sound quality, going from very bad (0) to very good (10). The subjective tests reveal a clear preference for the beamformer in controlled noisy situations as far as speech understanding is concerned. Results go in the same direction as those from the questionnaire based on the experience of the subjects at home, but the difference between the beamformer and hardware directional microphone is more pronounced. The average benefit of the beamformer for speech

understanding lies between 2 and 3.6 units for the different conditions. The benefit is larger for the single noise source scenario than for the multiple noise sources, which is in agreement with the results of the adaptive SRT measurements. As for sound quality, the CI subjects generally noticed less difference between the beamformer and the directional microphone: two subjects noticed no difference in sound quality; one subject rated BEAM on average 2.5 units better than the directional microphone. Two subjects judged listening with the beamformer more comfortable because of less background noise.

Subjective testing in real-life situations

The results of the subjective testing at home (evaluated by means of the SSQ questionnaire) are less pronounced than those from the subjective evaluation in the lab. A reason for this could be that, within the two-weeks the subjects evaluated the noise reduction programs at home, subjects did not spend a lot of time in noisy situations where the beamformer should lead to an advantage in speech understanding.

To test this hypothesis three subjects were taken out in real-life situations and again had to rate their perceived speech understanding in the vicinity of a speaker and the sound quality for both programs on an SSQ scale. Four situations were selected for this purpose: a ride by bus, a bus station, a shopping street and a pub/restaurant. The mean speech intelligibility benefit obtained with the beamformer goes from 0.3 to 1.2 on a score out of ten. The most important conclusion to be drawn from this test session is that also in real-life noisy conditions the subjective benefit from the beamformer is smaller than the reported benefit in controlled laboratory conditions. Possible explanations are that the real-life listening conditions were less noisy than the laboratory conditions. Noteworthy is also that the absolute scores for speech understanding with the directional microphone are 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. However, the difference between both programs in real-life is clearly present.

3. Localisation with bilateral hearing aids

The localization abilities in the horizontal plane have been evaluated with bilateral hearing aid users using different stimuli and different noise scenarios. The main cues for sound localization in these tests are interaural time differences (ITD) and interaural level differences (ILD). This study questions and quantifies the effect of current signal processing techniques on localization performance. The main research questions raised were (i) how do bilateral hearing aid users perform on a localization task, relative to normal hearing subjects, (ii) do modern bilateral hearing aids preserve localization cues and hence can hearing aid users use the full potential of their binaural processing

capabilities, (iii) is there an influence of state of the art noise reduction algorithms, more in particular an adaptive directional microphone configuration, on localization performance? The hearing aid users were tested without and with their hearing aids, using both a standard omnidirectional microphone and an adaptive directional microphone configuration. A group of normal hearing subjects was used as a reference. Low-frequency, high-frequency and broadband signals were used to separate the effects on ITD and ILD processing. Jammer sources were added to the condition with the broadband signal to evaluate the impact of the adaptive directionality system.

3.1 Methods

Subjects

Prior to the study with the hearing impaired subjects, as a reference a similar study was performed with ten young normal hearing subjects. Ten hearing impaired subjects, ranging from 44 to 79 years old, participated in this study. All of them are experienced bilateral hearing aid users. Six of them use Phonak Perseo hearing aids, three use GNResound Canta7 hearing aids and one of the test persons uses Widex Diva hearing aids. Monaural spectral enhancement techniques were switched off in the Canta7 and Perseo devices. The mean absolute difference between left and right hearing loss is less than 10dB for all subjects, except for subject ML. The subjects used their own earmolds with a venting between one and three mm, except for subject ML who used an open venting on her best ear for otological reasons. The amplification levels of all subjects did not show big asymmetrical settings (<7dB difference between the mean left and right amplification levels at an input level of 50dB SPL and 80dB SPL at all frequencies), except for subject ML who had asymmetrical amplification levels to compensate for the asymmetrical hearing loss. However, this subject did not show a bias with or without hearing aids and showed similar results as all other subjects.

Set-up

Tests were carried out in a reverberant room with dimensions 6m x 3m x 3.5m (length x width x height) and a reverberation time, T60, of 0.54s as determined for a speech weighted noise spectrum. Test persons were placed inside an array of 13 single-cone speakers with a cone diameter of 10cm. The speakers were located at a distance of 1 meter in the frontal horizontal plane at angles ranging from -90° to +90° relative to the subject, a spacing of 15° was used. The target signal was played through one of the 13 speakers using a LYNXONE soundcard and a programmable electronic switch box controlled by our ALP test software. Two powered speakers were placed at a distance of 1 meter from the subject at an angle of -90° and +90°.

Stimuli

Localization of high and low frequencies rely on different binaural processing strategies. To obtain information about both binaural processing paths a 200-ms, 1/3-octave, low-frequency noise band (fc=500Hz) and a 200-ms, 1/3-octave, high-frequency noise band were chosen as target stimuli. In the first study with normally hearing subjects, a 200-ms, 1/3-octave, high-frequency noise band centered at 5000Hz was used. Due to the inaudibility of the stimulus to some test persons, a 200-ms, 1/3-octave, high-frequency noise band centered at 3150Hz was used for the hearing impaired subjects. The third stimulus was a one-second broadband telephone ringing signal. It contains both low and high frequencies and includes a lot of transients. All target signals are cosine windowed with a rise and fall time of 50ms. The telephone stimulus was tested in silence as well as with a multitalker babble source located at the left and the right side of the subject. When tests were carried out with hearing aids or with normal hearing subjects, stimuli were presented at 65dB SPL. For the noise scenario, the two noise sources were set at 62dB SPL giving an SNR of 0dB in the center of the speaker array. Sound level calibrations were done in absence of the subject. To rule out the effect of audibility, tests were carried out at equal sensation levels with and without hearing aids. The amplification level of the stimuli was corrected until the subject confirmed that an equal sensation level was obtained with and without hearing. Afterwards, the noise level was corrected to keep the signal to noise ratio of 0dB. Because of the extra amplification in the unaided condition, the results do not reflect a "daily life" comparison between the aided and unaided condition. However, they reflect the ability of the subject to use binaural cues.

Test protocol

Subjects sat inside the array on a chair elevated until the ears reached the level of the speakers. Most tests (all tests with the hearing impaired and both tests with the telephone signal for the normal hearing) have three repetitions on each speaker resulting in 39 presented trials per test. The stimuli were presented randomly and roved with a roving level of 4dB. All subjects were instructed to keep their head fixed and pointed to 0° during stimulus playback.

They were watched on a monitor. Hearing aid users were tested without their hearing aids, with their hearing aids using the omnidirectional microphone and with their hearing aids using an adaptive directional microphone configuration. Two different acoustical scenarios were tested with the broadband stimulus (in silence and with multitalker babble sources). All 4 test scenarios were performed twice for each test subject. Two sessions were held on different dates for test-retest purposes and all tests were completed during each session.

The performance measures Root Mean Square Error (RMS) and Mean Absolute Error (MAE) have been used for analysis. The smallest non-zero error a subject can make

during one test run equals 2.4° RMS and 0.4° MAE (with n=39 trials per test run). Statistical analysis has been performed on both error measures, showing similar results. In this document statistical analysis will be shown only for RMS values.

3.2 Results and discussion

In figure 3 the group average results are presented as RMS and MAE for the different stimulus conditions for the normal hearing and the hearing impaired subjects. The naming of the different test conditions is as follows: the first letters represent the normal hearing group (nh) and the hearing impaired group with an omnidirectional microphone configuration (o), an adaptive directional microphone configuration (a) and without hearing aids (no). This is combined with the stimulus condition: the 500Hz stimulus (500), the 3150Hz and 5000Hz stimulus (3150 and 5000) and the broadband telephone ringing stimulus (t). The telephone signal was tested in silent conditions (ts) and with multitalker babble jammer sources placed at the left and right side of the subject (tblr). Only the averaged data over test and retest is shown because no significant difference was found between the test and retest condition for both normal hearing and hearing impaired subjects. The statistical analysis is done with test and retest data separated in a repeated measures ANOVA.

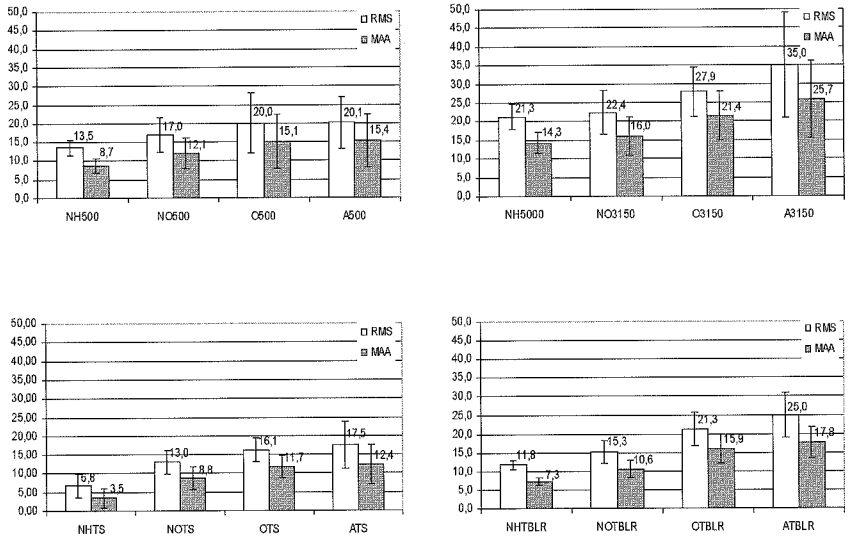

Figure 3
The average RMS and MAE error performance measures (in degrees) for the normal hearing and hearing impaired subjects in the study, for the different stimuli used.

The three research questions raised will be addressed.
Overall, the average performance with the low-frequency stimulus (mainly ITD) was better than performance with the high-frequency (mainly ILD) stimulus. This agrees with a trend seen in the work of Recanzone et al. (1998), although they used pure tone stimuli instead of noise bands. Test results improved when using the broadband stimulus compared to high or low-frequency stimuli. This can be explained by the possibility to use both ILD and ITD cues and by the time structure of the broadband signal. When adding jammer sources with a SNR of 0dB, performance drops significantly for both the hearing impaired subjects and the normal hearing subjects. This confirms the results of the study of Lorenzi et al (1999b).

It was also shown that for both the normal hearing subjects as the hearing impaired subjects, localization is more accurate in front of the listener than at the left and right sides of the listener which agrees with earlier research done on the minimal audible angle (Moore, 1997). However, the hearing impaired subjects using their own hearing aids, and are therefore highly trained on localizing sound sources with these devices in their daily lives, were not able to localize half of the targets correctly in the most frontal region whereas normal hearing subjects have a near 100 percent score in this region.

Statistical analysis showed a better performance of the normal hearing subjects compared to the hearing impaired subjects. Although the two subject groups were not age matched, 64 percent of the individual scores of the hearing impaired subjects without hearing aids are within two standard deviations of the results of the normal hearing group (when comparing the data of the same test conditions). In Lorenzi et al. (1999a), it was also mentioned that a considerable percentage of the hearing impaired subjects (between 2 and 3 subjects out of 4) can reach normal hearing performance on a binaural task. This concludes that some hearing impaired subjects are able to use binaural cues which is important to motivate further research on binaural signal processing for hearing aids.

To answer the question if bilateral hearing aids preserve localization cues, tests were performed with hearing impaired subjects with and without hearing aids. By using equal sensation levels (for restoring audibility) a comparison can be made between performance when all binaural information is present and performance with hearing aids. Throughout the different tests, 4 out of 8 comparisons between unaided and aided conditions showed significant better performance unaided than aided for a significance level of p=0.05 (and 6 out of 8 for a significance level of 0.1). A general analysis confirmed that performance without hearing aids is significantly better than with hearing aids. This can be related to the data of Byrne et al. (1998) obtained with open earmolds, allowing the subjects to use binaural cues of the direct sound instead of the output of the hearing aid. Moreover, when comparing with normal hearing performance, only 36 percent of the individual test results with hearing aids fall within 2 standard deviations of the

results of the normal hearing subjects. These percentages are considerably smaller than the numbers shown above for the hearing impaired subjects without hearing aids. This confirms that hearing aid users localize better without their hearing aids than with their hearing aids.

Extra measurements were made to quantify the influence of microphone position on interaural cues. On a BTE hearing aid, the microphones are positioned relatively far from the eardrum, which may influence the interaural cues given to the listener. To quantify this influence, physical measurements are made using the microphones of BTE hearing aid devices and in the ear microphones (ITE) of a 01dB CORTEX MK2 manikin under anechoic conditions. The BTE devices are two Canta7 dual microphone shells with direct microphone outputs. All recordings are made using an 8-channel G.R.A.S. 12AG pre-amplifier and 2 synchronized LYNXONE soundcards. A broadband white noise signal is recorded simultaneously with all 6 microphones. Interaural time delays (ITD's) and interaural level differences (ILD's) are calculated between the two in the ear microphones (ITE) of the manikin, the two omnidirectional microphones located at the front of the BTE devices and between the two omnidirectional microphones located at the back of the BTE devices. The data illustrate that the ITD's between the BTE microphone pairs do not fully agree with the data of the ITE microphone pair, which represents the ITD information at the eardrums of a human listener. Moreover, the distortion is dependent on the microphone placement on the BTE. The front microphone pair shows a distortion of ITD information only around -90° and +90°. The back microphone pair shows larger distortions and generates almost similar ITD's for the area from -60° to -90° and from 60° to 90° which could have an effect on localization performance.

In general both ITD and ILD distortions are relatively small, especially if one would assume that the hearing aid manufacturer uses the front microphone as the reference microphone. To rule out the effect of microphone placement the data is split into two parts: the data of the area where the impact of the microphone position is almost negligible (from -45° to 45°) and the remaining data for which the distortion could be (although unlikely) generated by the microphone position. The statistical analysis was repeated for these two subareas. A general analysis on the 4 test conditions confirms that performance in the area between -45° and 45° is worse with than without hearing aids. This suggests that the drop in localization performance cannot be explained only by microphone positioning. Because microphone positioning seems insufficient to explain the difference in localization performance, the data strongly suggests that the signal processing of hearing aids introduces interaural cue distortion when used in a bilateral hearing aid configuration.

Although overall no significant difference between the adaptive directional microphone configuration and the omnidirectional microphone configuration was shown (p=0.053), the adaptive directional microphone is significantly different from the omnidirectional microphone configuration when directional noise is added at +/-90° (p=0.044). The adaptive directionality has a negative impact only in this test condition, probably because the noise sources were playing continuously throughout this test. Therefore the noise reduction system had plenty of time to adapt to a steady filtering operation. In the other tests very short sounds were used which could have given the noise reduction system not enough time to adapt to a steady filtering operation. Two reasons might explain the negative effect of the adaptive directional microphone. Firstly, an adaptive directional microphone creates a directional pattern depending on the noise scenario. It searches for the most dominant noise source (normally the search area is restricted to the rear horizontal plane) and puts a null in this direction. When testing with babble jammer sources at the left and right side of the subject, the adaptive directional microphone will try to cancel out these directions. It can be assumed that the ILD perception of the stimulus around +/-90° could be degraded by this filtering operation. This would explain why the difference between the omnidirectional and the adaptive directional microphone configuration is only significant in the area from 60 to 90 degrees and not in the area from -45 to 45 degrees. Secondly, the directional filter introduces a phase relationship between in- and output of the hearing aid. ITD's would be distorted if left and right hearing aid have a different phase relationship.

These results indicate that an adaptive directional algorithm can have an influence on binaural cues. Therefore, when designing noise reduction systems for bilateral hearing aids, more attention should be given to the preservation of interaural cues.

4. Conclusions

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. Significant improvements of both the SRT in noise and the percentage correct phoneme scores were observed with BEAM compared to the standard hardware directional microphone. In addition, the SSQ questionnaire and the subjective evaluation in controlled and real-life scenarios showed a preference for the beamformer in noisy environments. This is the first adaptive noise reduction strategy actually implemented in a commercial CI.

Tests with bilateral hearing aid users using different stimuli and different noise scenarios demonstrate that: (i) bilateral hearing aid users perform worse than normal hearing subjects in a localization task, although more than half of the subjects reach normal hearing performance when tested unaided. For both groups localization performance drops significantly when acoustical scenario's become more complex. (ii) bilateral

independently operating hearing aids do not preserve localization cues (iii) overall, adaptive directional noise reduction can have an additional and significant negative impact on localization performance.

Acknowledgements
This research work was carried out in the frame of IWT-project 020540, FWO-project G.0233.01, a Ph.D grant of the Institute for the Promotion of Innovation through Science and Technology in Flanders (IWT-Vlaanderen), and was partially sponsored by Cochlear Ltd. The authors would also like to thank L. Royackers, M. Laureyns, K. De Klerck, R. Vangelder, K. Devoecht, J.-B. Maj for help during these studies.

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