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Comparison of adaptive noise reduction algorithms in dual microphone hearing aids ☆

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Abstract

In this paper, a physical and perceptual evaluation of two adaptive noise reduction algorithms for dual-microphone hearing aids are described. This is the first comparison between a fixed directional microphone on the one hand, and an adaptive directional microphone and an adaptive beamformer on the other hand, all implemented in the same digital hearing aid. The adaptive directional microphone is state-of-the-art in most modern commercial hearing aids. The *physical evaluation* shows the importance of an individual calibration procedure for the performance of the noise reduction algorithms with two microphone hearing aids. The directivity index calculated in anechoic conditions and intelligibility-weighted polar diagrams measured in reverberant conditions show that all the noise reduction strategies yield an improved signal-to-noise ratio (SNR), but that the adaptive beamformer generally performs best. From the *perceptual evaluation*, it is demonstrated that the adaptive beamformer always performs best in single noise source scenarios. In a more complex noise scenario, there is still a SNR improvement with all the techniques, however the effect is the same for all the strategies.
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Keywords: Adaptive beamformer; Adaptive directional microphone; Calibration; Noise reduction algorithms; Hearing aids

1. Introduction

Noise reduction strategies are important in hearing aid devices to improve speech intelligibility in a noisy background (Plomp, 1994). In commercial hearing aid devices, most often *fixed (software or hardware) directional microphones* (FDM) are used taking advantage of the spatial filtering effect of the dual-microphone system and reducing sound input from well-defined angles (Thompson, 1999). More recently, *adaptive directional microphones* (ADM) have been developed and implemented in hearing aids (Luo et al., 2002; Ricketts and Henry,

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2002). These systems can adapt to changing jammer (or interfering) sound directions and can track moving noise sources. Ricketts and Henry (2002) evaluated a fixed software directional microphone and an adaptive directional microphone for hearing aids, in a moderately reverberant condition ($T_{60} \approx$ 400–450 ms). The experiments demonstrated that the advantage of an adaptive over a fixed directional microphone was prominent when noise sources are situated on the side of the listener. With two noise sources at 75° and 105° (0° is the front direction), an improvement of the Speech Reception Threshold (SRT: defined as the sound-pressure level of speech at which 50% of the speech is correctly understood by the listener) of about 2 dB was obtained with the adaptive directional microphone relatively to an omnidirectional microphone. In this noise scenario, the fixed software directional microphone gave roughly the same SRT as the omnidirectional microphone. With noise sources at 165° and 195°, both directional microphones (fixed and adaptive) had roughly the same behaviour and a SRT-improvement of about 4 dB was obtained relatively to the omnidirectional microphone. In research, an extension of the Generalized Sidelobe Canceller (GSC) (Griffiths and Jim, 1982) was developed for dual microphone behind-the-ear (BTE) hearing aids. This extension of the GSC was called a *two-stage adaptive beamformer* (A2B) and an improvement of speech understanding in noise of 5 dB was obtained, relative to the effect of one fixed hardware directional microphone (VandenBerghe and Wouters, 1998). Other evaluations of this technique were carried out with normal hearing listeners and hearing impaired listeners (Wouters et al., 2002; Maj et al., 2004) as well as with cochlear implant patients (Wouters and VandenBerghe, 2001). In these studies, significant improvements of the speech intelligibility in noise were obtained. An average SRT-improvement about 7–8 dB was obtained for a single noise source at 90° for both normal hearing and hearing impaired listeners in a moderately reverberant condition ($T_{60} \approx 760$ ms) (Maj et al., 2004).

In this paper, the first comparison between a fixed software directional microphone, an adaptive directional microphone (Luo et al., 2002; Ricketts and Henry, 2002) and a two-stage adaptive beamformer (Maj et al., 2004) is carried out. A physical and a perceptual evaluation of the noise reduction algorithms are performed when all the strategies are implemented in the same hearing aid. The device is a commercial GNReSound Canta7 BTE hearing aid with two omnidirectional microphones and is chosen because it contains a free programmable DSP.

The paper is organized as follows. Section 2 described the concept of the three noise reduction techniques, the fixed directional microphone, the adaptive directional microphone and the adaptive beamformer. In Section 3, the experimental set-up for the physical and the perceptual evaluations is examined. The reverberant properties of the test room are describes and the hearing aid is presented. The latter has two omnidirectional microphones mounted in an endfire array configuration. Section 4 describes the performance measures and the different experiments of the physical evaluation. In a two-microphone configuration, the acoustical characteristics (magnitude and phase) differ for each microphone and this may have an impact on the performance of the noise reduction algorithms. In the physical assessment, the SNR at the output of a fixed directional microphone for five different hearing aids is measured with and without an individual calibration procedure. Polar diagrams and directivity indices are evaluated in anechoic and/or reverberant acoustical conditions. In Section 5, the test materials and the performance measure for the perceptual evaluation is presented. The latter is performed by 15 normal hearing subjects in four different noise scenarios and with two different types of jammer (or interfering) noise sounds. The improvement of the speech intelligibility is measured by evaluating the Speech Reception Threshold (SRT) of all strategies. In total, 608 SRT-measurements are carried out. In Section 6, the results of the physical and the perceptual evaluation are discussed. The physical evaluation shows that for having a similar performance, individualized matching or calibration of the fixed parts of the noise reduction strategies is necessary. The perceptual evaluation shows that the improvement in SNR with the two-stage adaptive beamformer technique, as well as with other noise reduction approaches, depends on the jammer sound scene. In simple jammer sound source scenarios the A2B always performs better than the fixed directional microphone and the adaptive directional microphone. However, in more complex noise scenarios, such as multiple jammer sound sources or diffuse noise, the performance of the adaptive algorithms falls back to the effect of the fixed directional microphone.

2. Noise reduction techniques

2.1. Fixed directional microphone

Based on hardware directional microphone strategy (Hawkins and Yacullo, 1984; Leeuw and Dreschler, 1991; Maj et al., 2004), fixed software directional microphones have been developed (Thompson, 1999; Bachler and Vonlanthen, 1995). The fixed software directional microphone (FDM) signal is obtained as the difference between the front omnidirectional microphone signal and a delayed version of the rear microphone signal (Fig. 1). The delay operation is implemented by a FIR (finite-impulse response) filter operation (w_1^{FDM}), which in our case has 10 filter-taps. The filter coefficients are optimized to obtain a hypercardioid polar diagram in anechoic conditions. The FIR filter operation is used to compensate the differences in magnitude and phase between the microphones of the hearing aids. This strategy differs from the conventional software directional microphone (Fig. 1) (Thompson, 1999), which has only one filter-tap and assumes that the characteristics of the microphones are matched. The additional delay operation on the signal of the front omnidirectional microphone enables the use of a non-causal filter, and its value is set to half of the size of the filter w_1^{FDM}.

Several studies show that the fixed directional microphone gives a SRT-improvement of about 3 dB in difficult listening conditions (Maj et al., 2004; Hawkins and Yacullo, 1984; Leeuw and Dreschler, 1991).

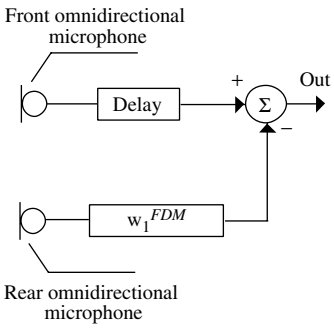


Fig. 1. Representation of the fixed directional microphone (FDM). The FDM signal is obtained as the difference between the front microphone and a delayed version of the rear microphone. The delay is implemented by a filter operation (w_1^{FDM}).

2.2. Adaptive directional microphone

The adaptive directional microphone (ADM) (Luo et al., 2002; Ricketts and Henry, 2002; Cezanne et al., 1995), is illustrated in Fig. 2 and is similar to what is the state-of-the-art in most modern commercial digital hearing aids, such as the Phonak Claro and GNReSound Canta7.

Two software directional microphones create reference signals, namely the *speech reference* and the *noise reference*. The speech reference is obtained as the difference between the front microphone signal and a delayed version of the rear microphone signal. The parameter $w_1(f)$ of the software directional microphone is the frequency-dependent weight for the back port $w_1(f) = a \cdot \mathrm{e}^{-\mathrm{j}2\pi f\tau}$, where the internal delay τ and the weight a are chosen to give a cardioid spatial characteristic in anechoic conditions. This delay is created by an all-pass filter operation. The speech reference is made with a front cardioid (e.g. null at 180°). The noise reference is similarly obtained and is made with a rear cardioid (e.g. null at 0°). The signals of the software directional microphones, the speech reference and the noise reference, are connected to an adaptive noise canceller (ANC). The FIR filter w_2^{ADM} of the ANC in our case has one tap and can be updated by means of classical adaptive algorithms (Haykin, 1996). Also, a constraint is applied on the coefficient of the ANC. This constraint can be seen as a voice activity detector (VAD) which is dependent on the spatial characteristics of the sound. This constraint allows the adaptation of the coefficient when a source is at the back

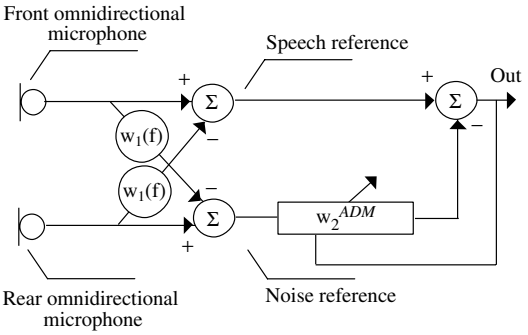


Fig. 2. Representation of the adaptive directional microphone (ADM) where the signals of the *speech reference* and the *noise reference* are created with two software directional microphones. The delay operation $w_1(f)$ allows to obtain a front cardioid (null at 180°) for the top branch and a rear cardioid (null at 0°) for the lower branch. The filter w_2^{ADM} of the ANC has one coefficient and is updated during processing.

hemisphere (e.g. 90°–270° where the source is considered as noise) and stops the adaptation when a source is at the front hemisphere (e.g. 270°–90° where the source is considered as speech). This avoids the cancellation of the speech signal at the output of the ANC (Cox et al., 1987).

2.3. Two-stage adaptive beamformer

The two-stage adaptive beamformer (A2B) is illustrated in Fig. 3 (VandenBerghe and Wouters, 1998). This algorithm is an extension of the Generalized Sidelobe Canceller (Griffiths and Jim, 1982) and was already evaluated with normal hearing listeners and hearing impaired listeners (Wouters et al., 2002; Maj et al., 2004) as well as with cochlear implant users (Wouters and VandenBerghe, 2001). However, these evaluations were carried out with the algorithm implemented on a real time DSP research platform separate from the BTE (DSP on PC-card or on Audallion platform).

A software directional microphone and a filter operation ($\mathbf{w}_1^{\mathrm{A2B}} = a \cdot \mathrm{e}^{-\mathrm{j}2\pi f\tau}$) are used to create the speech reference and the noise reference of the ANC. The delay is created by an all-pass filter operation. The software directional microphone has a hypercardioid spatial characteristic in anechoic conditions (null at 110°). The first filter is fixed and gives a look direction to the adaptive beamformer. It is assumed that the speaker is in front of the listener, i.e. at 0°. The delay operations effectively allow for a non-causal response of the first and the second filter, and their values are set to half of the size of the filters. The number of taps are 10 and 30, respectively for the first filter ($\mathbf{w}_1^{\mathrm{A2B}}$) and the filter of the ANC ($\mathbf{w}_2^{\mathrm{A2B}}$). The adaptive filter of the ANC uses a normalized least mean squares procedure (NLMS) (Haykin, 1996), and attempts to model noise during noise periods, and subtracts noise from speech-plus-noise, when speech is present. A VAD algorithm is implemented to decide whether the signal contains speech-plus-noise or noise only. The VAD used in this study is an extension of the VAD developed by Van Gerven and Xie (1997) and is based on the log-energy of the directional microphone. For more details about the implementation of the two-stage adaptive beamformer, the reader is referred to (Maj et al., 2004; Maj et al., 2004).

3. Experimental set-up

In this study, a physical and perceptual evaluation of the fixed directional microphone, the adaptive directional microphone and the two stage adaptive beamformer was carried out. The noise reduction algorithms were implemented in GNReSound Canta7 hearing aids and the output signals of the different algorithms as well as the front omnidirectional microphone signal were available at the loudspeaker of the hearing aids based on a selection program. For both evaluations, the choice of the program and parameter settings were done from a PC-platform, with software from Audiologic and GNReSound, through a transmission device (HI-PRO). For the physical and the perceptual evaluation, the same reverberant room was used. This room had a volume of 70 m^3 ($2.8 \times 3.5 \times 7.1$ m). The reverberation time (T_{60}) of the room was obtained for 12 different positions of a loudspeaker and for an unmodulated speech-weighted noise signal. The mean of T_{60} for the 12 positions (and the

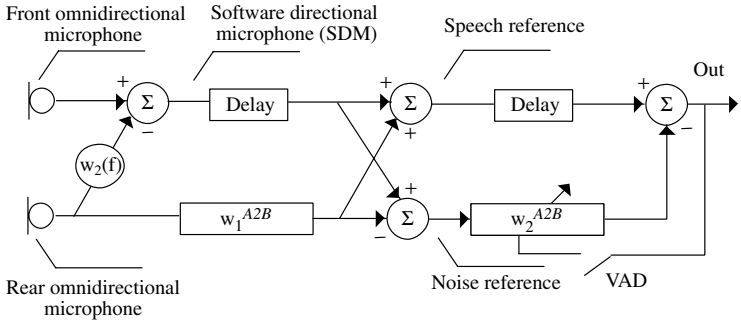


Fig. 3. Representation of the two stage adaptive beamformer (A2B) where a software directional microphone and a filter operation, namely first filter (w_1^{A2B}), are used to create the signals of the speech and the noise reference of the ANC. The delay operation of $w_2(f)$ allows to obtain a front spatial hypercardioid (null at 110°). The first filter is fixed and gives a look direction to the adaptive beamformer, the angle 0°.

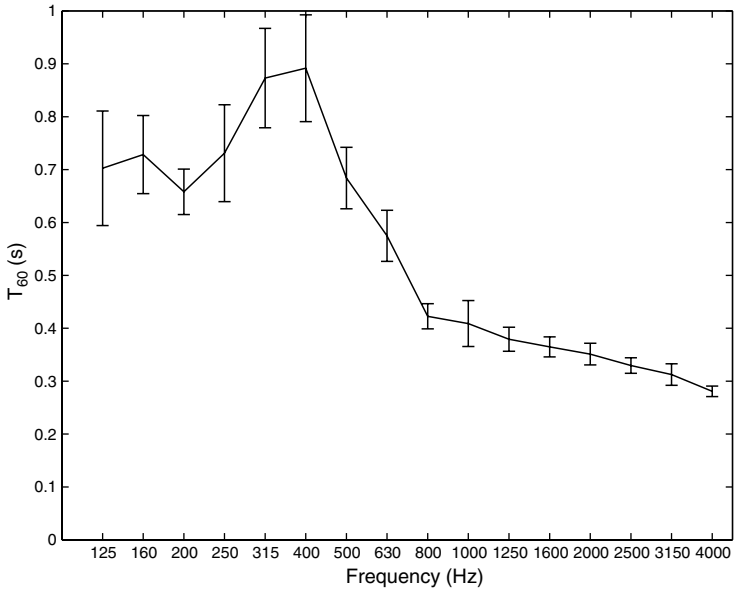


Fig. 4. Reverberation time (T_{60}) of the reverberant room was calculated for 12 different positions of a loudspeaker for an unmodulated speech weighted noise. The means and the standard deviations for 16 center frequencies from 125 Hz to 4 kHz of the one-third octave band are represented. With the full-band speech weighted noise, an average of 0.76 s is obtained for the T_{60}.

standard deviation) is presented in Fig. 4 for 16 center frequencies from 125 Hz to 4 kHz of one-third octave bands. With the full-band speech weighted noise, an average T_{60} of 0.76 s was obtained and is representative of moderately reverberant environments which are representative of real living rooms. The critical distance, which is the distance where the power in the reverberant sound field equals the power in the direct sound field, was measured with a sound level meter (Rion NA-27) in one-third octave bands. The critical distance was about two meters over the average of 0.5, 1 and 2 kHz. A reference point in the room was defined where, for physical measurements, the center of the head of the mannequin and, for perceptual the measurements, the center of the head of the subjects were positioned. All loudspeakers were at one meter from this reference point.

3.1. Hearing aid

The noise reduction algorithms were implemented for behind-the-ear (BTE) GNReSound Canta7 hearing aids (Fig. 5). This platform is a powerful free programmable digital platform. Two omnidirectional microphones (Microtronic-9667GX1) are mounted in an endfire array configuration spaced 1.6 cm apart. For these tests, a linear

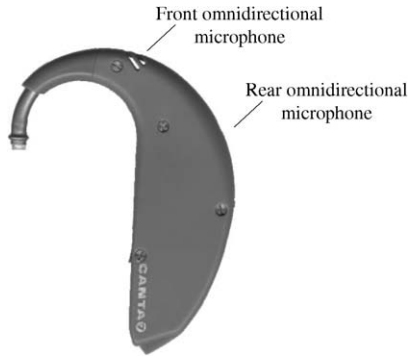


Fig. 5. GNReSound Canta7 hearing aid where two omnidirectional microphones are mounted in an endfire array configuration spaced 1.6 cm apart.

amplification was used in the hearing aid and the systems for compression or feedback control were switched off. Five Canta7 hearing aids, as obtained from the fabrication line, were used in this study.

3.2. Physical assessment

For the physical evaluation, the loudspeaker of the hearing aid was connected to an ear simulator coupler (Bruel & Kjaer 4157) mounted on a microphone with a preamplifier (Bruel & Kjaer 4165 and Bruel & Kjaer 2639). The signal of this microphone

was amplified by a Bruel & Kjaer (2610) amplifier and was digitized, at a sampling frequency of 16 kHz, with a PC-platform using a LynxOne sound card with 16-bit analog-to-digital conversion. The test signals were sent through a loudspeaker located at one meter in the azimuth plane of the hearing aid (in stand-alone) or from the center of the head of the mannequin. The signals at the loudspeaker of the hearing aid were recorded for different locations of the loudspeaker, corresponding to angles between 0° (front direction endfire angle) and 360° in steps of 15°. The level at the loudspeaker was adjusted to obtain a sound level of 65dBSPL at the reference point, with the head removed. The physical measurements were carried out in an anechoic and in the reverberant test room (described above). The latter was also the same as used for the perceptual evaluation.

3.3. Perceptual assessment

For the perceptual evaluation, five loudspeakers (Yamaha CBX-S3) with approximately flat (±2 dB for 150 Hz–10 kHz) frequency response were used for the speech and discrete noise sources in the test room: one for the speech signal in front the subject at 0°, and four for presenting noise signals at 45°, 90° (side of the hearing aid), 180° and 270°. The loudspeakers were calibrated separately to obtain the same sound level at the reference point for each loudspeaker. The middle of the loudspeakers was at the same height as the reference point (140 cm) and the distance between each loudspeaker and the middle of the subject's head was one meter. For noise scenarios with more than one spectrally identical jammer noise, the presented signals were uncorrelated. In the control room, a second room, the speech signals were output by a computer sound card (Sound blaster 16 bits) and the noise signals by SONY CD991 CD players. For amplification purposes and intensity control, the speech signal was sent through a MADSEN OB822 audiometer (channel right), the noises at 45° or 90° (depending on the noise scenario) through the MADSEN OB822 audiometer (channel left) and the noise at 180° and 270° through an AMPLAID 309 audiometer (channel left and right).

4. Physical evaluation

The physical evaluation involved acoustic measurements in an anechoic chamber with the hearing aid in stand-alone configuration or mounted on a mannequin, to calibrate the filters of the noise reduction algorithms and to calculate a directivity index (DI). Also acoustic measurements were performed in a reverberant chamber (which is described above) when the hearing aids are mounted on a mannequin to calculate intelligibility-weighted polar diagrams.

4.1. Calibration

The noise reduction algorithms need a calibration for two different purposes, namely to match the level of the microphone signals at 1.6 kHz (central frequency-band important for speech intelligibility) and to calculate the coefficients of the fixed filters. The latter defines the spatial characteristics of the fixed directional microphone and the first filter of the adaptive beamformer. The calibration was carried out in an anechoic chamber with five different Canta7 hearing aids.

To match the level of both microphones of the hearing aid at 1.6 kHz, a gain was applied to the rear omnidirectional microphone signal. To calculate this gain, the hearing aid was in stand-alone configuration and the loudspeaker (Yamaha CBX-S3) located at 90° (relative to the endfire angle) at one meter in the azimuth plane of the hearing aid. The test signal was a white noise filtered by a one-third octave band filter with center frequency at 1.6 kHz. An ANC algorithm with one coefficient updated by means of a NLMS procedure was used to determine the gain. The adaptation of the ANC was stopped after a few seconds and the obtained coefficient was used as the gain for the rear microphone. With the Audiologic software, the adaptation of the coefficients of the adaptive filters could be checked.

The fixed directional microphone had a hypercardioid polar diagram (null at 110°) and the coefficients were defined to minimize the output signal when a source was present at the angle 110°. For this, a test signal was presented by a loudspeaker located at an angle 110° in the azimuth plane of the center of the mannequin. The test signal was based on the multilanguage long-term average speech spectrum (LTASS) from the 'ICRA' compilation (International Collegium of Rehabilitative Audiology). The ICRA-signal was an unmodulated noise representative of a male weighted idealized speech spectrum at normal effort (ICRA, 1997). The filter coefficients were adapted, by means of a

NLMS procedure and the adaptation was stopped after a few seconds. The obtained coefficients were then used to create the fixed directional microphone. The coefficients of the first filter of the adaptive beamformer were defined to give a specific look direction to the algorithm, more specifically, the coefficients were calculated to minimize the presence of the speech signal in the noise reference. The same procedure as for the fixed directional microphone was performed but now with the loudspeaker at 0°.

4.2. Physical performance

In anechoic conditions, the evaluation was performed with the ICRA-signal (described above). In reverberant conditions, a stationary speech-weighted noise was used, corresponding to Dutch sentences from a male speaker (also used in the perceptual evaluation). The signals of the omnidirectional microphone and the fixed directional microphone were directly accessible through the ear simulator set-up. The signal of the ADM and the A2B changes with the adaptation of the second filter. To carry out the physical evaluation of the adaptive algorithms, the coefficients were adapted on the noise signal until convergence and then they were kept fixed. Then, the reference signal (at 0°) and noise signals were filtered separately with the fixed filters. These two separately filtered signals were then used for the physical evaluation of the adaptive directional microphone and the adaptive beamformer.

The DI (Beranek, 1954) is an often-used performance measure for directional microphone configuration and noise reduction schemes in hearing aids (Desloge et al., 1997; Ricketts, 2000; Ricketts, 2000). It has already been shown that the DI has a strong link with the prediction of the improvement of the speech intelligibility in noise (Ricketts, 2000). With θ the azimuth coordinate and ϕ the elevation coordinate, the directivity index for a hearing aid in stand-alone configuration and in anechoic conditions can be expressed as

$$\mathrm{DI}(f) = \frac{4\pi|P(f,0,0)|^2}{\int_0^{2\pi}\int_0^{\pi}|P(f,\theta,\phi)|^2|\sin\theta|\,\mathrm{d}\theta\,\mathrm{d}\phi} \tag{1}$$

where the $|P(f,\theta,\phi)|^2$ is the mean squared sound pressure level, at frequency f, of the output signal of the hearing aid when the sound source is located at the coordinate (θ,ϕ). If symmetry is assumed in the vertical plane and there is reasonable symmetry around the horizontal plane, the DI can be calculated from only the $|P(f,\theta,\phi=0)|^2$ values recorded at discrete angles of the horizontal plane by using the following formula (Beranek, 1954; Ricketts, 2000):

$$\mathrm{DI}(f) = \frac{8\pi 57.3^\circ|P(f,0,0)|^2}{2\pi\sum_{i=1}^{180^\circ/\delta\theta}|P(f,\theta_i,\phi=0)|^2|\sin\theta_i|\delta\theta} \tag{2}$$

where f are the 16 center frequencies from 160 Hz to 5 kHz of the one-third octave bands and $\delta\theta = 15^\circ$. An intelligibility-weighted version of the DI can be defined as

$$\mathrm{DI}_{\mathrm{AI}} = \sum_{i=1}^{k} I_i \mathrm{DI}_i \tag{3}$$

where I_i is the weight for the importance of the ith one-third octave band (Pavlovic, 1987).

Peterson et al. (1989) and Greenberg et al. (1993) used the articulation index (ANSI, 1969) procedure to develop an intelligibility weighted SNR. Since 1997, an extension of the articulation index calculation has been suggested. This extension is known as the speech intelligibility index (SII) (ANSI, 1997), where the weights for the audibility function A_i to some extent take into account a variety of adverse listening conditions, such as noise masking, filtering, distortion and low reverberation. In this study, the polar diagrams show the intelligibility-weighted SNR as a function of the direction of the source (i.e. azimuth). The intelligibility-weighted SNR is defined as

$$\mathrm{SNR}_{\mathrm{SII\ weighted}} = \sum_{i=1}^{k} I_i A_i \mathrm{SNR}_i \tag{4}$$

where SNR_i is the signal-to-noise ratio measured (in dBSPL) in the ith one-third octave band. I_i and A_i are the weights for the importance of the band and the audibility function, respectively, as described by the speech intelligibility index SII (ANSI, 1997). The weights A_i were calculated in accordance with the SII procedure for one-third octave bands and for the standard speech spectrum level at the raised vocal effort (68.3 dBSPL) (ANSI, 1997).

5. Perceptual evaluation

The tests with the three strategies (FDM, ADM and A2B) were carried out in four different noise scenarios (single noise source at 45°, at 90°, at 180° and three uncorrelated noise sources at 90°/

180°/270° relative to speaker position). Two spectro-temporally different noise sounds (SW: unmodulated stationary speech weighted noise and MB: multitalker babble) and one speech material (sentences spoken by a male speaker) were used. Because of the calibration issues raised all perceptual measurements were carried out with the same Canta7 hearing aid. In total, 480 tests were carried out and 128 test–retest evaluations were performed.

5.1. Subjects

Fifteen normal hearing listeners participated in these measurements. Their ages ranged from 19 to 23 years (mean of 21 years) and their pure tone thresholds were less than or equal to 20 dBHL at the octave frequencies from 125 Hz to 8 kHz. The perceptual evaluation was carried out in a monaural configuration and the hearing aid gain was 0 dB. The monaural listening situation was created by an earplug (Bilsom 303S-30 or 303L-30) and by an earcap (Bilsom 2301) at the non-test ear.

5.2. Test materials

To measure the SRT, an adaptive procedure was used (Plomp and Mimpen, 1979). The adaptive procedure adjusts the level of the speech material to extract the 50% speech recognition level, i.e. the SRT in a noise background at fixed level. The first sentence of a speech list was repeated with increasing level until the subject correctly identified it. Once this level was determined, every other sentence of the list was subsequently presented only once at a level lower or higher, depending on the former item being identified correctly or not. The level step size for each speech material item was 2 dB. In order to determine the SRT, the levels of the last 10 responses of a list of thirteen sentences were averaged. The four noise sources were calibrated separately, to obtain a constant sound level at the reference point. Thus, a sound level was obtained of 65 dBSPL and 69.8 dBSPL with one noise source and three noise sources, respectively. Sentences spoken by a male speaker were used as speech materials, and, unmodulated speech weighted noise (SW: with identical spectrum as the corresponding speech material) and multitalker babble (MB) were used as noise materials. The sentence speech materials were the Dutch sentences developed by Versfeld et al. (2000). These sentences are an extension of the materials of Plomp and Mimpen (1979) to measure speech reception thresholds. Thirty nine lists were available, and each list contained thirteen sentences. One of the two test noises was unmodulated noise speech weighted according to the spectrum of the specific speech materials used. The other noise was multitalker babble, taken from the compact disk Auditory Tests (Revised) edited by Auditec of St. Louis.

6. Results and discussion

6.1. Physical evaluation

The calibration of the noise reduction algorithms was performed for five different hearing aids. To check the influence of the calibration on the noise reduction performance, the intelligibility-weighted polar diagrams of the fixed directional microphone were measured in stand-alone and anechoic condition. In a first experiment, the calibration of the fixed directional microphone was performed with one hearing aid and this calibration was kept for the four other hearing aids. Fig. 6 shows the intelligibility-weighted polar diagrams of the five hearing aids and it can be seen that there were important variations in the shape of the polar diagrams. The non-smooth curves for the four hearing aids (slim line on Fig. 6) is due to the 15° step size of the measurement procedure. The differences between the maximum and the minimum of the $SNR_{SII\ weighted}$ for the five hearing aids are 7.5 dB, 2.9 dB and 9.2 dB at angles 105°, 180° and 255°, respectively. In a second experiment, the calibration of the fixed directional microphone was carried out for the five different hearing aids separately. The intelligibility-weighted polar diagrams of the five hearing aids are depicted in Fig. 7. The shapes of the polar diagrams are seen to be roughly similar. In this case, the differences between the maximum and the minimum of the $SNR_{SII\ weighted}$ for the five hearing aids are 1.9 dB, 1.1 dB and 3.3 dB at angles 105°, 180° and 255°, respectively. This means that the differences in magnitude and phase between each microphone have an impact on the noise reduction performance of the algorithm. For optimal use, a calibration of every algorithm has to be performed for every hearing aid. The performance of the noise reduction algorithm depends too much on the set of microphones used in the hearing aid, in particular on the level of matching between the microphones. From now on, only one hearing aid is used for the physical and the perceptual evaluations.

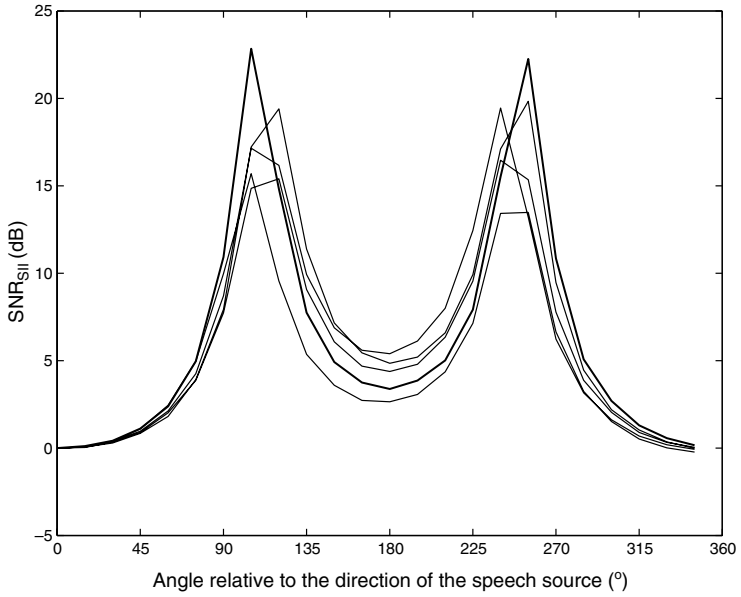


Fig. 6. Intelligibility-weighted polar diagrams (in dB) of the fixed directional microphone for five different hearing aids. The calibration of one hearing aid was carried out (thick line) and this calibration was kept for the four other hearing aids. The plotted values are relative to the angle of the direction of the speech source (0°), when the SNR is 0 dB, measured on the basis of speech noise in stand-alone configuration and in anechoic conditions.

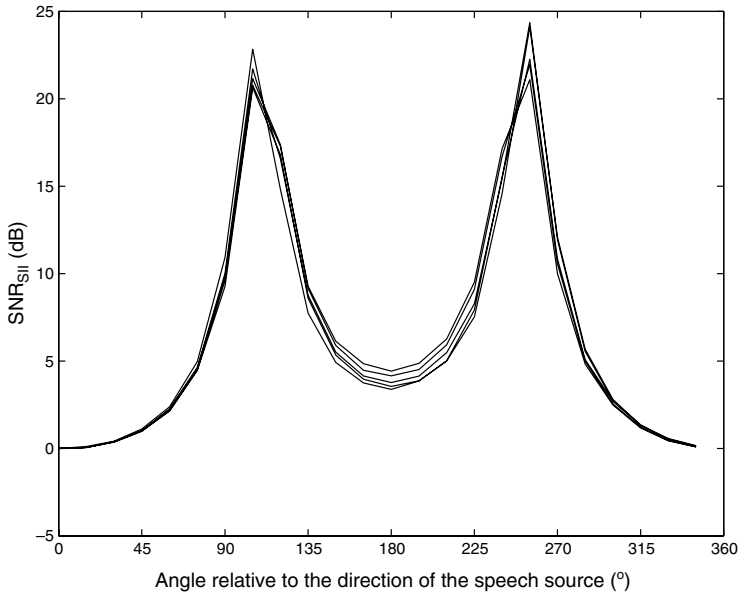


Fig. 7. Intelligibility-weighted polar diagrams (in dB) of the fixed directional microphone for five different hearing aids. The calibration was carried out separately for the five different hearing aids. The plotted values are relative to the angle of the direction of the speech source (0°), when the SNR is 0 dB, measured on the basis of speech noise in stand-alone configuration and in anechoic conditions.

The DI of each algorithm is presented in Fig. 8. As expected, the omnidirectional microphone (shown as reference here) has the same sensitivity for all angles resulting in a DI approximately 0 dB. At low frequencies (125–250 Hz), the noise reduction schemes have low values for the DI. The ADM performed worse than the omnidirectional microphone below 170 Hz. A negative DI means

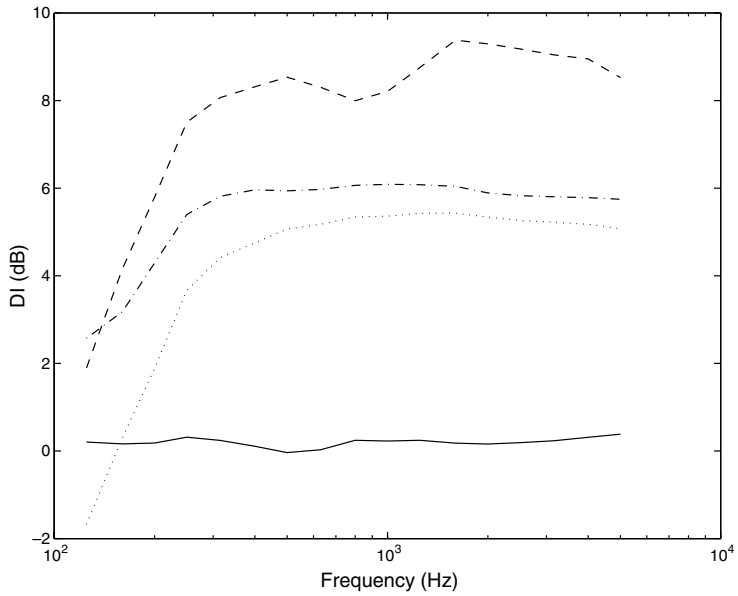


Fig. 8. The directivity index DI (in dB) of the omnidirectional microphone (—), the fixed directional microphone (-·-·-), the adaptive directional microphone (···) and the two stage adaptive beamformer (---) are presented. The plotted values are relative to the angle of the direction of the speech source (0°), when the SNR at the reference point is 0 dB, measured on the basis of speech noise and the hearing aid stand-alone configuration and in anechoic conditions.

that the algorithm is more sensitive to the back hemisphere than to the front hemisphere. Above 170 Hz, the A2B performed better than the FDM, which performed better than the ADM. Above 250 Hz, the A2B performed at least 2 dB higher than the FDM. The difference between the FDM and the ADM strategy was only 1 dB for frequencies above 450 Hz. The DI_{AI} equaled 0.2 dB for the omnidirectional microphone, 5.9 dB for the FDM, 5.1 dB for the ADM and 8.6 dB for the A2B.

Fig. 9 shows the intelligibility-weighted polar diagrams of the noise reduction algorithms and the front omnidirectional microphone in reverberant conditions. The interpretation of this polar diagram can be separated in three different hemispheres. A first on the side of the hearing aid between 0° and 105° (ipsilateral), a second at the back between 105° and 255° and a third on the contralateral side between 255° and 0°. In the ipsilateral hemisphere, the noise reduction algorithms always performed better than the omnidirectional microphone. The FDM had roughly the same effect as the ADM. The A2B performed better than the other noise reduction algorithms especially between 45° and 90°. This improvement was at least 2 dB higher at 45° up to 4 dB at 90°. In the back hemisphere, the ADM and the A2B technique had roughly the same performance except for angles from 210° to 255° where the A2B was slightly better than the ADM algorithm. The main difference in this hemisphere was between the adaptive algorithms and the FDM. At 180°, the adaptive schemes performed about 4.4 dB better than the fixed directional microphone. On the contralateral hemisphere, the four algorithms had roughly the same performance in SNR. The A2B was always slightly better than the FDM and the ADM techniques but the improvement did not exceed 1 dB.

6.2. Perceptual evaluation

Table 1 shows the improvements (in dB) of the SRT relative to the omnidirectional microphone for the fixed microphone (FDM) and the two adaptive techniques (ADM and A2B). The data correspond to the mean (and the standard deviation) of these improvements of all 15 subjects for the four noise scenario configurations and the two noise materials.

An analysis of variance (ANOVA) was carried out using SPSS 10.0 statistical analysis software. In what follows, the *F*-value is a measure of the distance between the distributions of the SRT-data obtained in different test conditions, and the *p*-value is the probability that the data belong to the same

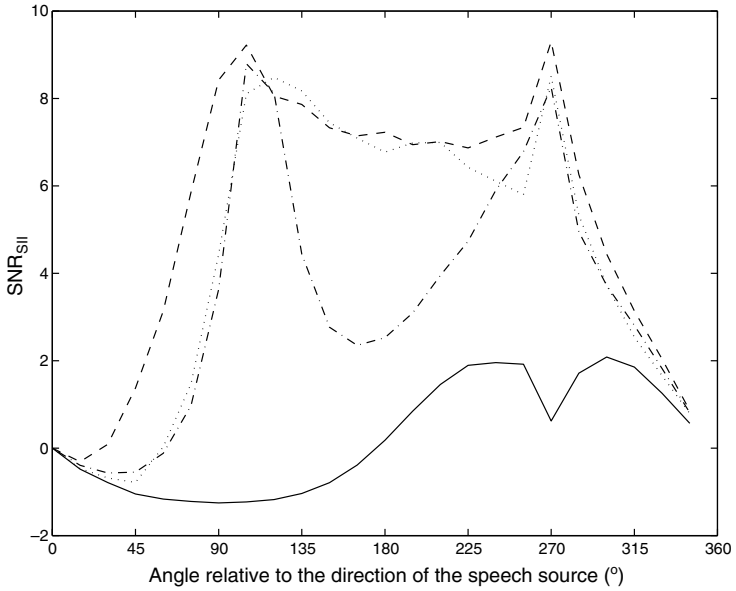


Fig. 9. Intelligibility-weighted polar diagrams (in dB) of the omnidirectional microphone (—), the fixed directional microphone (-·-·), the adaptive directional microphone (···) and the two stage adaptive beamformer (---). The angles are relative to the direction of the speech source (0°). The SNR at the center of the head is 0 dB, measured on the basis of speech noise, with the hearing aid on a mannequin and in reverberant conditions ($T_{60} = 0.76$ s).

Table 1
The differences in SRT (dB) averaged for the 15 normal hearing listeners (mean (SD)) of the fixed directional microphone, the adaptive directional microphone and the adaptive beamformer, relative to the omnidirectional microphone, for different test conditions

Noise configuration		FDM	ADM	A2B
45°	SW	0.8 (2.3)	0.2 (1.9)	2.2 (2.0)
	MB	1.9 (3.8)	−0.2 (2.8)	2.6 (2.4)
90°	SW	4.7 (3.3)	3.6 (2.8)	8.1 (3.2)
	MB	5.0 (3.9)	3.5 (3.5)	8.4 (2.9)
180°	SW	3.2 (3.0)	4.7 (3.3)	6.9 (2.7)
	MB	3.3 (3.8)	5.9 (4.1)	7.6 (3.8)
45°/90°/180°	SW	3.1 (2.8)	2.8 (2.8)	4.4 (2.5)
	MB	4.3 (3.3)	3.2 (3.4)	3.9 (2.7)

statistical distribution. For p-values smaller than 0.05 the test conditions are considered to yield significantly different results (Hinkle et al., 1998). The tests on within-subject analysis showed that there were significant differences between the noise materials ($F = 508.8$, $p < 0.001$), between the noise scenarios ($F = 47.3$, $p < 0.001$) and between the noise reduction algorithms ($F = 49.1$, $p < 0.001$). From tests of within-subject contrasts, the analysis showed that the interaction term between the algorithms and the noise materials was not significantly different from chance ($F = 0.369$, $p = 0.776$). This means that the noise reduction performance of the different techniques was independent of the noise materials. This allowed us to average the SRT improvements obtained with the stationary speech-weighted noise and multitalker babble noise tests. However, the absolute SNR at 50% intelligibility for the speech material (or SRT) was on average about 7.3 dB higher for the multitalker babble than for the stationary speech weighted noise. Speech understanding in the multitalker babble noise was significantly more difficult than in stationary speech weighted noise. For each noise reduction technique, the improvements in SRT, relative to the omnidirectional microphone, depended on the noise scenarios. The performance of the noise reduction scheme was smallest when the noise source was close to the speaker, at 45°. With the FDM (4.8 dB) and the A2B (8.2 dB), the best performance was obtained with a noise source at 90°, and the ADM (5.3 dB) performed best with a noise source at 180°. The four first subjects carried out all the SRT measurements two times in two different days. This was done to obtain an idea about the within-subject variability of the obtained results. The mean difference between both tests was 3.3 dB with a standard deviation of 4.2 dB. From a separate t-test analysis, it was shown that the first and the second measurement were not

significantly different ($p = 0.489$). To compare the performance of the noise reduction techniques, a paired comparison was carried out for the different noise configurations.

With a noise source at 45°, there were no significant differences between the signal of the omnidirectional microphone and the signals of the FDM ($p = 0.072$) and the ADM ($p = 0.977$). Only the A2B performed significantly different from the omnidirectional microphone ($p < 0.001$). However, there were no significant differences between the FDM and the A2B ($p = 0.167$). Relatively to the omnidirectional microphone, an improvement in SRT of 1.3 dB was found with the FDM and 2.4 dB with the A2B. No improvement in SRT was obtained with the ADM. These results were in close agreement with the physical evaluation (Fig. 9).

With a noise source at 90°, all the noise reduction techniques performed significantly different from the omnidirectional microphone ($p < 0.001$). The largest improvement in SRT was found with the A2B, namely 8.2 dB relatively to the omnidirectional microphone and 4.8 dB with the FDM. These results agreed with results from a previous study (Maj et al., 2004) where similar improvements in SRT were obtained for the A2B (8.7 dB) and for the FDM (4.7 dB) for identical conditions (speech material, noise material and test room). With the ADM, an improvement of 3.5 dB was obtained. There were no significant differences between the FDM and the ADM in a speech-weighted noise ($p = 0.084$). This was predicted by the physical evaluation on Fig. 9 at 90°. The behaviour of the two techniques was roughly the same. However, in multitalker babble there were significant differences between these two noise reduction techniques ($p = 0.018$). As already mentioned (Maj et al., 2004; Ricketts, 2000), intelligibility-weighted polar diagrams as well as the directivity index have a strong link with the improvement of the speech intelligibility in noise. The FDM performed better than the ADM in this noise scenario because the noise source was next to 110°, the angle where the FDM has a null. The improvement in SRT (3.5 dB) obtained by the ADM corresponded to the values measured for a software directional microphone in other studies (Maj et al., 2004). It seems that with the constraint on the adaptive filter (see Section 2.2), a noise source at 90° did not fully allow the adaptation of the adaptive filter. The software directional microphone (which was connected to the speech reference of the algorithm and had a null at 180°) was obtained at the output of the ADM.

For a single noise source at 180°, all the strategies were different from each other ($p < 0.001$). The A2B (7.2 dB) gave the best improvement in speech intelligibility. An improvement of 3.2 dB and 5.3 dB was obtained with the FDM and the ADM, respectively. The ADM was better than the FDM approach in this noise scenario. The noise source was not next to the optimal angle (110°) for the FDM and the noise source was at the back hemisphere, which means that the constraint allowed the adaptation in the ADM. In this way the adaptation part brought an additional noise reduction to the software directional microphone of the ADM.

With three uncorrelated noise sources, the noise reduction techniques all performed significantly different from the omnidirectional microphone ($p < 0.001$). However, there were no significant differences between the noise reduction algorithms. Improvements in SRT of 3.7 dB, 3.0 dB and 4.1 dB were obtained with the FDM, the ADM and the A2B, respectively.

In a single noise source scenario, the perceptual experiments revealed that the two stage adaptive beamformer always performed better than the fixed directional microphone and the adaptive directional microphone, which is state-of-the-art in most modern commercial hearing aids. This is certainly due to the complexity of the noise reduction algorithms. The adaptive beamformer needs more computation power than the other adaptive noise reduction technique, but its implementation is feasible in modern commercial hearing aids. The differences between the fixed directional microphone and the adaptive directional microphone depended mainly on the noise scenario. Indeed, this difference depended on the angle between the noise source and the optimal nulling angle of the fixed directional microphone (110°) and if the noise source was at the front or the back hemisphere for the ADM.

In a complicated noise scenario, the speech understanding in noise is clearly enhanced by the different techniques but that the adaptive systems had the same effect as the FDM, i.e. the adaptive schemes did not bring a significant additional SRT-improvement in complex noise scenarios. This additionally stresses the approach, as confirmed by data (Maj et al., 2004; Greenberg and Zurek, 1992), that a good noise reduction scheme should have an adaptive processing for low reverberation or simple noise source scenarios and a fixed microphone directivity for high reverberation or complex noise scenarios. The improvements in speech intelli-

gibility, as measured here, with the noise reduction algorithms in a complicated noise scenario, are very important for hearing-aid users. Indeed, in critical listening conditions (close to 50% of speech understood by the listener) an improvement of 1 dB in SNR can correspond to an increase of speech understanding of about 15% in every day speech communication (Plomp and Mimpen, 1979).

In hearing aid application, the robustness of the noise reduction strategies against microphone mismatch and imperfections in microphone mounting is an important issue. Indeed, the performance of the noise reduction techniques can be severely affected by the characteristics (magnitude and phase) of the microphones (Section 6.1) (Maj et al., 2004). In this study, the FDM and the A2B present the advantage over the ADM of having a calibrated filter operation. This calibration allow to compensate the differences in magnitude and phase between the two microphones of the hearing aids, unlike the first stage of the ADM which is created with conventional software directional microphones (Thompson, 1999). The latter strategy assumes that the characteristics of the microphone are matched.

The advantage of the ADM and the A2B over the FDM is the adaptive filter operation. The latter allows the noise reduction techniques to track moving noise sources. However in this study, the constraint applied on the adaptive filter of the ADM did not allow adaptation of the adaptive filter when a source was at the front hemisphere (270°–90°), and was maybe to restrictive. Therefore, the A2B seems to be a good compromise in terms of robustness against the characteristic of the microphone and adaptation to changing jammer (or interfering) sound directions, relative to the FDM and the ADM strategies.

Finally, the SRT-improvements measured in this study were performed within the direct sound field of the loudspeaker and in moderately reverberant conditions ($T_{60} = 0.76$ s) (Section 3). A reduced benefit to the important gain in SRT obtained in this study can be expected due to a shorter critical distance or increased reverberation.

7. Conclusion

Three dual-microphone noise reduction approaches were compared in this study. A physical evaluation demonstrated that individual matching or calibration of the two microphones is necessary for optimal performance. The physical experiments revealed that the two stage adaptive beamformer always performed better than the fixed directional microphone and the adaptive directional microphone, which is state-of-the-art in most modern commercial hearing aids. The perceptual experiments demonstrated that the SRT-improvements for the three noise reduction strategies depended on the jammer sound source scenario. The higher the complexity of the jammer sound scene, the lower the SRT-improvements. In a simple noise source scenario, the A2B brought a clear benefit in speech intelligibility relative to the FDM and ADM. However, in complicated noise scenario, the performance of the A2B fell back to the effect of the fixed directional microphone.

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