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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 a perceptual evaluation of two adaptive noise reduction algorithms for digital hearing aids is described. This is the first comparison between a fixed microphone pattern, an adaptive directional microphone and an adaptive beamformer, which are implemented in the same hearing aid.

1. INTRODUCTION

Noise reduction strategies are important in hearing aid devices to improve speech intelligibility in a noisy background [1]. Modern digital hearing aids using dual-microphone configurations in a single behind-the-ear (BTE) hearing aid allow more complex noise reduction algorithms. In commercial hearing aid devices, the most often used method is directional microphones taking advantage of the spatial filtering effect of the dual-microphone system by reducing sound input from well-defined angles [2]. More recently, adaptive algorithms were developed and implemented in hearing aids [3, 4]. These algorithms can adapt to changing jammer sound directions and can track moving noise sources. In this study, a physical and a perceptual evaluation of a fixed microphone pattern, an adaptive directional microphone and an adaptive beamformer is performed. These strategies are implemented in the same hearing aid, which is a commercial GNReSound Canta7 BTE hearing aid with two omnidirectional microphones. This hearing aid was chosen because it contains a free programmable DSP. In the physical part, polar diagrams and directivity index are evaluated in anechoic and/or reverberant acoustical conditions. Perceptual tests are carried out in an acoustical environment comparable to living room listening condition. The perceptual evaluation is performed by 15 normal hearing subjects in 4 different noise scenarios and with 2 different types of jammer noise sounds.

2. NOISE REDUCTION ALGORITHM

2.1. Fixed microphone pattern

The fixed directional microphone (FDM) is obtained as the difference between the signal of the front omnidirectional microphone and the delayed version of the rear microphone. The delay operation is implemented by a filter operation, which has ten coefficients. The coefficients are optimized to get a hypercardiod polar diagram in anechoic conditions.

2.2. Adaptive directional microphone

The adaptive directional microphone (ADM), similar to what is the state-of-the-art in the most modern commercial digital hearing aids, such as Phonak Claro and GNReSound Canta, is depicted in figure 1 [3]. Two software directional microphones create reference signals, namely the *speech reference* and the *noise reference*. The *speech reference* is made with a front cardioid (e.g. null at 180^o) and the *noise reference* is made a rear cardioid (e.g. null at 0^o). The signals of the software directional microphones, *speech and noise reference*, are connected to an adaptive noise canceller (ANC). The filter w_2^{ADM} of the ANC has one tap and can be updated by means of classical adaptive algorithms [5]. Also, a constraint is applied on the coefficient of the ANC. This constraint allows the adaptation of the coefficient when a source is at the back hemisphere (e.g. 90^o-270^o where the source is considered as noise) and stop the adaptation when a source is at the front hemisphere (e.g. 270^o-90^o where the source is considered as speech). This avoids the cancellation of the speech signal at the output of the ANC.

2.3. Two-stage adaptive beamformer

The adaptive beamformer (A2B) is depicted in figure 2 [4]. A software directional microphone and a filter operation (w_1^{A2B}) are used to create the signals of the *speech reference* and the *noise reference* of the ANC. The software directional microphone has to the front direction a hypercardioid (null at 135^o). The first filter is fixed and gives a look

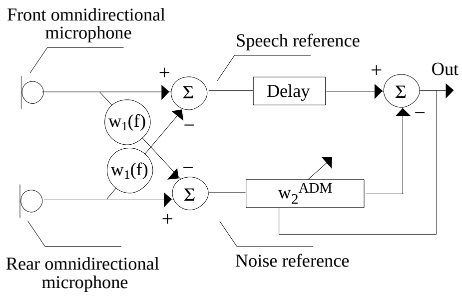

Fig. 1. Adaptive directional microphone.

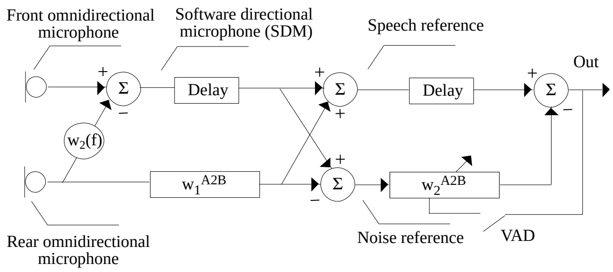

Fig. 2. Adaptive beamformer.

direction to the adaptive beamformer. It is assumed that the speaker is in front the listener, at about the angle 0^o. The delay operation at the signal of the software directional microphone allows having a non-causal response of the first filter. The sizes of the filters are 10 and 30 coefficients respectively for the first filter (w_1^{A2B}) and the filter of the ANC (w_2^{A2B}). The adaptive filter of the ANC uses a normalized least mean squares procedure (NLMS) [5], and attempts to model noise during noise periods, and subtracts noise from speech-plus-noise, when speech is present. A voice activity detector (VAD) algorithm is implemented to decide whether the signal contains speech-plus-noise or noise only.

3. HEARING AID

The noise reduction algorithms are implemented in a behind-the-ear (BTE) hearing aid which is a GNReSound Canta7 hearing aid. This platform is a very powerful free programmable digital platform available. Two omnidirectional microphones (Microtronic-9667GX1) are mounted in end-fire array configuration spaced 1.6cm apart. For these tests, a linear amplification is used in the hearing aid and the systems for compression or feedback control are switched off.

4. PHYSICAL EVALUATION

The physical evaluation involves acoustic measurement in an anechoic chamber with the hearing aids in stand-alone configuration to calculate a directivity index (DI) of the noise reduction algorithm. Acoustic measurements are also performed in a reverberant chamber when the hearing aid is mounted on a mannequin, to calculate intelligibility weighted polar diagrams. The test room has a reverberation time of 0.76s for a speech-weighted spectrum, this agrees well with the reverberation in typical living rooms. The DI [6] is an often-used measure of the performance of a directional microphone configuration and noise reduction schemes in hearing aids. It has already been shown before that the DI has a strong link with the prediction of the improvement of the speech intelligibility in noise [7]. With θ the azimuth coordinates and ϕ the elevation coordinates, the directivity index equals

$$DI(f) = 10.log\left(\frac{4\pi|P(f,0,0)|^2}{\int_0^{2\pi}\int_0^{\pi}|P(f,\theta,\phi)|^2.|sin\theta|.d\theta.d\phi}\right) \tag{1}$$

where the $|P(f,\theta,\phi)|^2$ is the magnitude of the mean squared sound pressure, 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 [6]:

$$DI(f) = 10.log\left(\frac{8.\pi.57.3^o.|P(f,0,0)|^2}{2\pi\sum_{i=1}^{180^o/\delta\theta}|P(f,\theta_i,\phi=0)|^2.|sin\theta_i|.\delta\theta_i}\right) \tag{2}$$

where f are the 16 center frequencies from 160Hz to 5000Hz of the one-third octave bands.
The polar diagrams show the intelligibility weighted signal-to-noise ratio SNR which is defined as:

$$SNR_{SIIweigthed} = \sum_{i=1}^{k} I_i.A_i.SNR_i \tag{3}$$

where SNR_i is the signal-to-noise ratio measured (in dB-SPL) in the i-th 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 [8]. 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 (68.3dBSPL) [8]. This speech spectrum level was chosen in function of the sound power level of the speech signals during the recordings (70dBSPL).

5. PERCEPTUAL EVALUATION

The perceptual evaluation is performed by 15 normal hearing listeners by measuring the Speech Reception Threshold (SRT) with an adaptive procedure [1]. The tests of the omnidirectional microphone, the fixed directional microphone, the adaptive directional microphone and the adaptive beamformer are carried out in 4 different noise scenarios (single noise source at 45^o, 90^o, 180^o and 3 independent noise sources at $90^o/180^o/270^o$ relative to speaker position 0^o). Two spectro-temporally different noise sounds (SW: unmodulated speech weighted noise and MB: multitalker babble) are used and one speech material, sentences from a male speaker [9].

6. RESULTS

The DI of each algorithm is presented in figure 3 as a function of frequency. As expected the omnidirectional microphone has the same sensitivity for all angles with the DI approximately 0dB. At low frequencies (125-250Hz), the three noise reduction schemes have low values of the DI. Above 170Hz, the adaptive beamformer performs better than the fixed scheme, which performs better than the adaptive directional technique. Above 250Hz the adaptive beamformer performs at least 2dB higher than the fixed technique. Figure 4 shows the intelligibility-weighted SNR, based on the SII, of the three noise reduction algorithms and the front omnidirectional microphone in reverberant conditions. The A2B performs better than the other noise reduction approaches especially between the angles 45^o and 90^o. This improvement is at least 2dB higher at 45^o up to 4dB at 90^o. Between the angle 105^o and 255^o, the adaptive techniques have roughly the same performance in SNR. The main difference in this hemisphere is between the adaptive algorithms and the fixed algorithm. At the angle 180^o, the adaptive schemes perform about 4.4dB better than the fixed directional microphone. Between the angle 255^o and 0^o, the three algorithms have roughly the same performance in SNR.

Table 1 shows the improvements (in dB) of the SRT relative to the omnidirectional microphone for the fixed directional microphone, the adaptive directional microphone and the adaptive beamformer respectively. The data correspond to the mean (and the standard deviation SD) of these improvements of all 15 subjects and is presented for the 4 noise configurations and the 2 noise materials. To compare the performance of the noise reduction techniques between each other, a statistical analysis (a paired comparison) is performed for the different noise configurations. With a noise source at 45^o, there are no significant speech intelligibility differences between the omnidirectional microphone and the FDM ($p=0.072$) or ADM ($p=0.977$). Only the SRT improvement with the A2B is significantly different from the omnidirectional microphone ($p<0.001$).

With a noise source at 90^o, the biggest improvement in SRT is obtained with the A2B, 8.2dB relatively to the omnidirectional microphone, 4.8dB with the FDM and 3.5dB with the ADM . There are no significant differences between

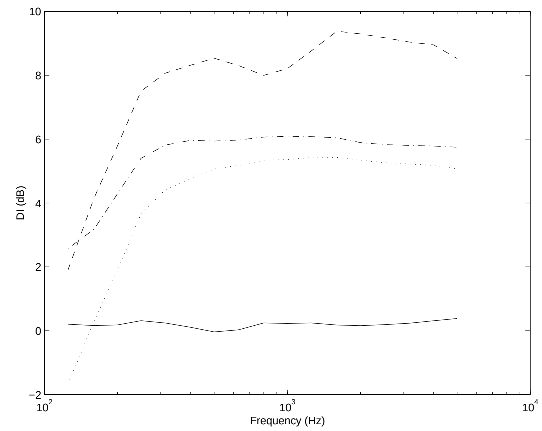

Fig. 3. The Directivity Index DI (in dB) of the omnidirectional microphone $(-)$, the fixed directional microphone $(-\cdot)$, the adaptive directional microphone $(\cdots)$, the adaptive beamformer $(--)$ are presented. The plotted values are relative to the angle of the direction of the speech source (0^o), when the SNR at the center of the head is 0dB (anechoic).

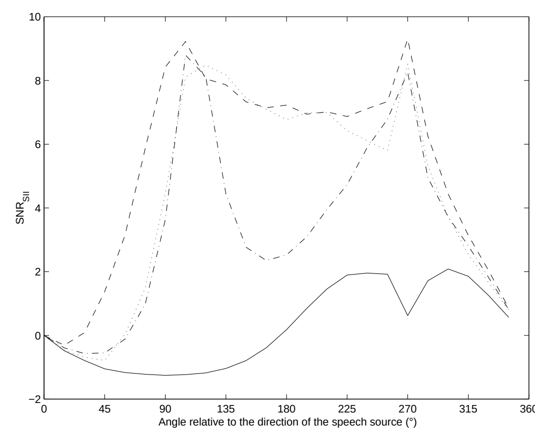

Fig. 4. Intelligibility-weighted SNR as a function angle for a single noise source (in dB) of the omnidirectional microphone $(-)$, the fixed directional microphone $(-\cdot)$, the adaptive directional microphone $(\cdots)$ and the adaptive beamformer $(--)$. The plotted values are relative to the angle of the direction of the speech source (0^o), when the SNR at the center of the head is 0dB (reverberant condition).

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 the 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)

the fixed directional technique and the ADM in a speech-weighted noise (p=0.084). However, in multitalker babble there are significant differences between these two noise reduction techniques (p=0.018). The FDM performs better than the ADM in this noise scenario because the noise source is next to 110^o, the angle where the FDM has a null. With a noise source at 180^o, all are independent of each other ($p<0.001$). The A2B (7.2dB) gives the best improvement in speech intelligibility. The ADM is better than the FDM in this noise scenario (3.2dB and 5.3dB respectively). The noise source is at the back hemisphere and the adaptation part of the ADM brings an additional noise reduction.
In a complicated noise scenario, with three noise sources, the signals of the noise reduction techniques are all significantly different from the signal of the omnidirectional microphone ($p<0.001$). However, there are no significant differences between the three different noise reduction algorithms. This means that a noise reduction is obtained by the different techniques but the adaptive systems have the same effect as a fixed microphone pattern. The adaptive schemes for 2 microphones hearing aids do not bring a significant additional SRT-improvement in complex noise scenarios. This additionally stresses the approach, as confirmed by data that a good noise reduction scheme should have an adaptive processing for low reverberation or simple noise scenarios and fixed processing for high reverberation or complex noise scenarios.

7. CONCLUSIONS

The adaptive beamformer always performs better or equal than the fixed directional microphone and the adaptive directional microphone. This is due to differences in complexity of the noise reduction algorithms. More coefficients are used in the filters of the adaptive beamformer than the adaptive directional microphone. The adaptive beamformer uses more computation power than the fixed and adaptive directional microphone, but its implementation is feasible in most recent commercial digital hearing aids. The differences in outcome between the fixed directional microphone and the adaptive directional technique depend mainly on the noise scenario. Indeed, this difference depends on the angle between the noise source and the optimal nulling angle of the fixed directional microphone (110^o) and if the noise source is at the front or the back hemisphere for the adaptive directional microphone. [1]

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