A TWO-STAGE ADAPTIVE BEAMFORMER FOR NOISE REDUCTION IN HEARING AIDS

Jean Baptiste Maj[1,2], *Jan Wouters*[1] *and Marc Moonen*[2]

[1]Lab.Exp.ORL KULeuven, Kapucijnenvoer 33, 3000 Leuven
[2]ESAT-SISTA KULeuven, Kasteelpark Arenberg 10, 3001 Leuven
Jean-Baptiste.Maj@uz.kuleuven.ac.be

ABSTRACT

In this paper, an evaluation is performed of a noise reduction algorithm for hearing aids using two microphones, based on two-stage adaptive beamforming. Theoretical, physical and perceptual evaluations are performed. A significant improvement of the signal-to-noise ratio (SNR) and the speech reception threshold (SRT) is obtained. With one noise source comes at an angle of 90^o, an improvement of 11.0dB and 8.2dB is obtained for the SNR and the SRT respectively, between the front omnidirectional microphone and the output of the two-stage adaptive beamformer.

1. INTRODUCTION

In [1], Vanden Berghe developed a noise reduction algorithm for Behind-The-Ear (BTE) hearing aids using two microphones referred to as a two-stage adaptive beamformer. The input signals to the algorithm were generated by two hardware directional microphones and the two stages of the beamformer used adaptive filters. An improvement in speech intelligibility of 5.5dB was obtained between a hardware directional microphone and the output of the two-stage adaptive beamformer. In this paper, an alternative to the scheme of Vanden Berghe is presented. For the sake of robustness, the filter in the first stage of the beamformer is kept fixed and for reasons of hardware implementation omnidirectional microphones are used. Theoretical, physical and perceptual evaluations of the two-stage adaptive beamformer are performed. These evaluations are carried out for speaker in front of the listener, at an angle of 0^o, and for a jammer noise source at 90^o on the side of the hearing aid. Theoretical and physical evaluations have shown that the processing of the two-stage adaptive beamformer does not distort the speech signal and improves significantly the SNR relatively to the omnidirectional microphone. Perceptual evaluations have shown that a significant improvement of the speech reception threshold (SRT: threshold where 50% of the speech is understood) is obtained with hearing impaired listeners.

2. TWO-STAGE ADAPTIVE BEAMFORMER

The two-stage adaptive beamformer has three different signal processing parts (figure 1). In the first part, where a software directional microphone (Dir) is created by using a fixed beamformer technique. In the second part, a filter W_1 is fixed to give a specific look direction to the two-stage adaptive beamformer. In practice, this filter is trained in anechoic conditions with the direction of the desired signal at 0^o [2]. Finally,

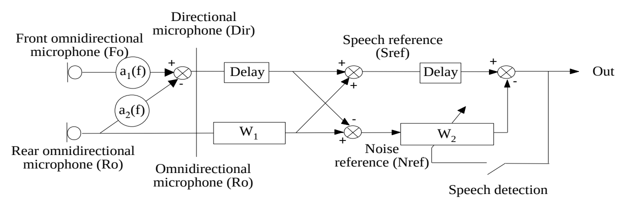

Figure 1: *Scheme of the two-stage adaptive beamformer.*

the third part implements an adaptive noise canceller (ANC), attempts to model noise during noise periods, and subtracts noise from speech plus noise when speech is present. A speech detection scheme is used. The sum and subtraction (middle part of figure 1) improves the noise reference (Nref) of the ANC. In practice, The number of coefficients are 10 and 30 for the first and the second filter, respectively. The additional delays actually allow to have non-causal filters, and their values are set to half of the size of the filters (5 and 15). The second filter is an adaptive filter and the coefficients are updated by a Normalized-Least Mean Square procedure (NLMS).

3. THEORETICAL PERFORMANCE

In the sequel, $P_{X,X}(f)$ is the power spectral density (PSD) of signal $X(f)$ and $P_{X,Y}(f)$ the cross-PSD of signals $X(f)$ and $Y(f)$ in the frequency domain.
The theoretical performances measures namely, noise

reduction $NR(f) = P^{noise}_{out,out}(f)/P^{noise}_{in,in}(f)$, speech conservation $SC(f) = P^{speech}_{out,out}/P^{speech}_{in,in}$ and noise sensitivity $\Psi(f) = P^{white}_{out,out}/P^{white}_{in,in}$, can be computed by using the complex coherence function (CCF) [3]. The CCF between two microphones is expressed by:

$$\Gamma_{Fo,Ro}(f) = \frac{P_{Fo,Ro}(f)}{\sqrt{P_{Fo,Fo}(f).P_{Ro,Ro}(f)}} \tag{1}$$

With a source situated at angle θ, and when sensor noise is present, the CCF becomes:

$$\Gamma_{Fo,Ro}(f) = \frac{exp(j.2.\pi.f.cos(\theta).d/c)}{1+\rho(f)} \tag{2}$$

Where $\rho(f)$ is the sensor-to-environmental noise ratio, d the distance between the two microphones and c the velocity of the sound in air ($c \approx 340m/s$).
The PSD at the output of the beamformer $P_{Out,Out}$ can be expressed by the CCF and the PSD of the signals in different part of the beamformer as follows (see figure 1).
• PSD at the directional microphone:

$$\begin{aligned} P_{Dir,Dir}(f) = P_{in,in}(f).(&\sum_{i=1}^{2} a_i(f).a_i^*(f) \\ &+ 2.Re(a_1(f).a_2^*(f).\Gamma_{Fo,Ro}(f)) \end{aligned} \tag{3}$$

$$P_{Dir,Ro}(f) = P_{in,in}(f).(a_1(f).\Gamma_{Fo,Ro}(f) + a_2(f)) \tag{4}$$

where $P_{in,in}(f)$ denotes the PSD of the speech signal $P^{speech}(f)$ or the noise signal $P^{noise}(f)$ ($P^{noise}_{Ro,Ro}(f) = P^{noise}_{Fo,Fo}(f) = P^{noise}_{in,in}(f)$ and $P^{speech}_{Ro,Ro}(f) = P^{speech}_{Fo,Fo}(f) = P^{speech}_{in,in}(f)$).
• PSD at the first stage:

$$\begin{aligned} P_{Sref,Sref} = &P_{Dir,Dir}(f) + |W_1(f)|^2.P_{Ro,Ro}(f) \\ &+ 2.Re(W_1^*(f).P_{Dir,Ro}(f)) \end{aligned} \tag{5}$$

$$\begin{aligned} P_{Nref,Nref}(f) = &P_{Dir,Dir}(f) + |W_1(f)|^2.P_{Ro,Ro}(f) \\ &- 2.Re(W_1^*(f).P_{Dir,Ro}(f)) \end{aligned} \tag{6}$$

$$\begin{aligned} P_{Sref,Nref}(f) = &P_{Dir,Dir}(f) - |W_1(f)|^2.P_{Ro,Ro}(f) \\ &+ 2.Im(W_1^*(f).P_{Dir,Ro}(f)) \end{aligned} \tag{7}$$

The first filter W_1 is kept fixed and equals:

$$W_1(f) = \frac{P^{speech}_{Dir,Ro}(f)}{P^{speech}_{Ro,Ro}(f)} \tag{8}$$

• PSD at the output of the two-stage adaptive beamformer:

$$\begin{aligned} P_{out,out}(f) = &P_{Sref,Sref(f)} + |W_2(f)|^2.P_{Nref,Nref}(f) \\ &- 2.Re(W_2^*(f).P_{Sref,Nref}(f)) \end{aligned} \tag{9}$$

The second filter is adapted during noise periods and is given by:

$$W_2(f) = \frac{P^{noise}_{Sref,Nref}(f)}{P^{noise}_{Nref,Nref}(f)} \tag{10}$$

The noise sensitivity is defined as the ratio of the spatially white noise gain to the gain of the desired signal [3]. It is used to quantify the sensitivity to model and processing errors. The $P^{white}_{out,out}(f)$ is found by calculating $P_{out,out}(f)$ when $\Gamma_{Fo,Ro}(f) = 0$. Figures 2 and

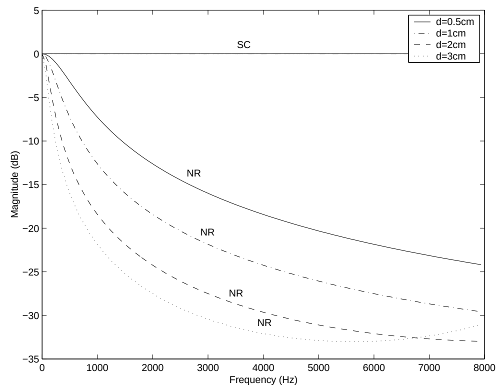

Figure 2: Influence of the distance between the two microphones on the $SC(f)$ and the $NR(f)$ of the two-stage adaptive beamformer.

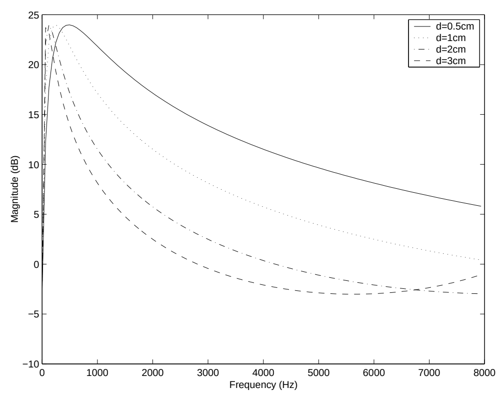

Figure 3: Influence of the distance between the two microphones on the noise sensitivity $\Psi(f)$ of the two-stage adaptive beamformer.

3 show the influence of the distance between the two microphones on the $SC(f)$, the $NR(f)$ and $\Psi(f)$ performances. Independently of the sensor noise and the distance between the two microphones, the $SC(f)$ has the same performance. The transfer function is flat and equals 0dB, which means that the speech signal is not distorted by the processing of the beamformer. On the other hand, the distance between the microphones influences the $NR(f)$ and the $\Psi(f)$ performance measures. The shorter the distance, the worse the performance. Figure 4 shows that the sensor noise also influences the $NR(f)$ performance. The higher the sensor noise, the worse the $NR(f)$ performance. Hence, by decreasing the distance between the microphones, the algorithm becomes more sensitive to the sensor noise and reduce the $NR(f)$ of the algorithm.

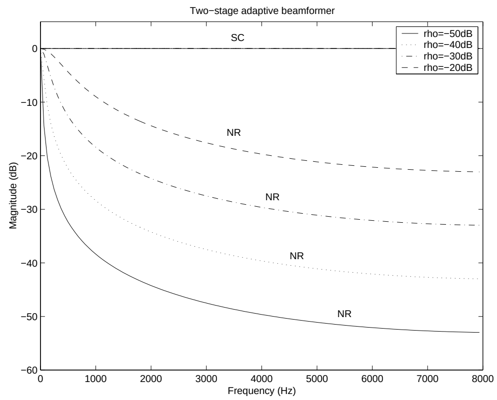

Figure 4: Influence of the sensor noise on the speech conservation $SC(f)$ and the noise reduction $NR(f)$ $(d = 2cm)$.

4. PHYSICAL EVALUATION

For the physical evaluation, the used hearing aid is a Danavox-163D hearing aid housing with two omnidirectional microphones mounted in an endfire array configuration spaced two centimeters apart. A speech-weighted noise [4] was presented 1 meter in front of a dummy head and at angles in steps of 15^o. It implies that the noise and speech signals have the same spectrum. The reverberation time of the test room was $T_{60} = 0.76s$. The SNR relative to the case where the noise source is at angle 0^o, is measured at the omnidirectional, directional microphone and the output of the beamformer (figure 5). The omnidirectional microphone has the same sensitivity for all angles (around 0dB in our case), however, the sensitivity is seen to be a function of the angle, mainly due to the effect of the dummy head. The output of the two-stage adaptive beamformer always give a larger improvement than the omnidirectional and the directional microphone. Remarkably, the directional microphone can perform worse than the omnidirectional microphone, namely between angles 0^o and $+60^o$. At 90^o, figure 5 shows that a SNR improvement of 8dB between the output of the beamformer and the directional microphone, while an improvement of 3dB between the directional and the omnidirectional microphone is measured. For this angle, the transfer functions between the directional microphone and the output of the two-stage adaptive beamformer signals and between the front omnidirectional microphone and the output of the beamformer signals for speech and noise are shown figure 6.

The transfer function between the directional microphone and the output of the beamformer for the speech signal is around 0dB and corresponds to the theoretical analysis. Hence, the two-stage adaptive beamformer processing does not distort the speech signal. However, the transfer function between the omnidirectional microphone and the output of the beamformer indicates an attenuation at the low frequencies (under 1500Hz). This is due to the characteristics of the microphones, which have a cut-off in the low frequencies. Hence, the directional microphone attenuates the low frequencies. This attenuation did not appear in the theoretical analysis, where the characteristics of the microphones were not taken in account. Otherwise, above the frequency 1500Hz, the transfer function corresponds to the theoretical analysis.

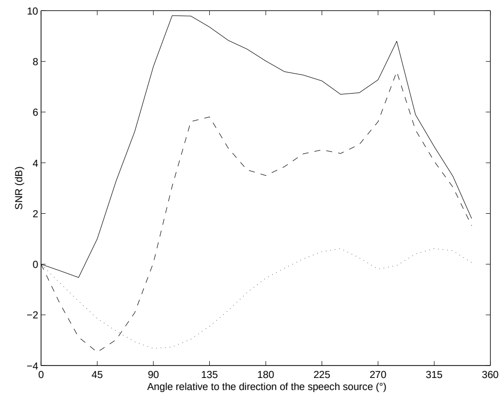

Figure 5: The curves show the SNR (in dB) at the output of the two-stage adaptive beamformer $(-)$ as a function of noise source position angle relative to the direction of the speech source, at the directional microphone $(--)$ and at the omnidirectional microphone $(\cdots)$.

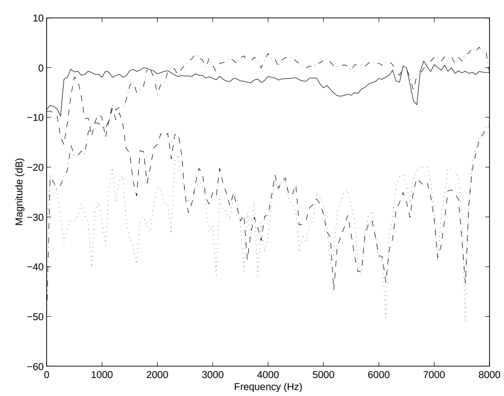

Figure 6: $(-\cdot)$ shows the transfer function for a speech signal and $(\cdots)$ for a noise signal between the front omnidirectional microphone and the output of the two-stage adaptive beamformer. $(-)$ shows the transfer function for a speech signal and $(--)$ for a noise signal between the directional microphone and the output of the two-stage adaptive beamformer.

5. PERCEPTUAL EVALUATION

Perceptual tests have been performed with 5 hearing impaired listeners. The SRTs has been measured for

3 conditions: omnidirectional microphone, directional microphone and output of the beamformer, using an adaptive method [2]. These measurements were done in the same conditions as the physical analysis (loudspeaker, position of the loudspeaker, hearing aid and configuration of the room). At 0^o, speech sentences (spoken by male and female voices) were presented, and at 90^o a speech-weighted noise with the same long-term average spectrum as the sentences, as well as multitalker babble were presented at a level of 65dBSPL.
For the tests, the hearing aid was connected to a digital signal processor (DSP, MOTOROLA-56009) in the Audallion. The Audallion is a wearable DSP research platform that allows performing clinical trials. The sampling rate is 15.625 kHz. The patient wore the hearing aid and was sitting between the loudspeakers, with the head situated at 1 meter of the speech and the noise source.
Table 1 shows the SRT measurements. A statistical analysis (ANOVA) has shown that there are significant differences between the speech materials and noise sound used between test conditions and the three signals. However, the differences between the three conditions were not significantly different for the different speech and noise test materials used. An average improvement of 8.2dB with a standard deviation (STD) of 3.5dB is found between the omnidirectional microphone and the output of the beamformer, and, an improvement of 5.1dB with a STD of 2.8dB is found between the directional and the omnidirectional microphone. The big STD of the measurements are caused by differences in the position of the hearing aid behind the ear, the hair of the patient (long and short hair), the position of the head of the patient, etc.

6. CONCLUSIONS

A significant SRT improvement of hearing impaired listeners is obtained with the directional microphone and the output of the two-stage adaptive beamformer relatively to the omnidirectional microphone. These improvements are important for hearing-aid users, because in critical listening conditions (close to 50 per cent of speech understood by the listener) an improvement of 1dB in SNR corresponds to an increase of speech understanding of about 15 per cent in every day speech communication [5]. [1]

Sp. Mat.	Prog.	P1	P2	P3	P4	P5
Man Sent. in SW	Fo	71.8	68.2	69.0	68.6	69.8
	Dir	66.2	63.8	63.8	67.0	63.8
	Out	61.8	60.6	63.0	65.8	60.6
Impr.	Fo−Dir	5.6	4.4	5.2	1.6	6.0
	Fo−Out	10.0	7.6	6.0	2.8	9.2
Woman Sent. in SW	Fo	73.8	69.8	68.2	69.4	69.8
	Dir	67.0	65.4	65.0	66.2	61.0
	Out	65.0	62.2	59.4	67	56.2
Impr.	Fo−Dir	6.8	4.4	3.2	3.2	8.8
	Fo−Out	8.8	7.6	8.8	2.4	13.6
Man Sent. in bab.	Fo	76.6	76.2	75.0	76.2	78.2
	Dir	72.6	73.0	71.0	72.2	66.6
	Out	70.6	68.2	68.6	69.0	63.8
Impr.	Fo−Dir	4.0	3.2	4.0	4.0	11.6
	Fo−Out	6.0	8.0	6.4	7.2	14.4
Woman Sent. in bab.	Fo	78.6	72.6	72.6	76.6	75.0
	Dir	70.6	70.2	72.6	69.4	66.2
	Out	67.8	68.6	66.2	68.6	59.0
Impr.	Fo−Dir	8.0	2.4	0.0	7.2	8.8
	Fo−Out	10.8	4.0	6.4	8.0	16.0

Table 1: SRT of the patients (Px) for sentences spoken by male and female in a stationary speech-weighted noise of the sentences (SW) and multitalker babble (bab) at 65dBSPL. All values in dB The improvements in SRT are also shown.

7. REFERENCES

[1] Vanden Berghe, J. Wouters, J.: *An Adaptive Noise Canceller for Hearing Aids Using Two Nearby Microphones*. The Journal of the Acoustical Society of America, 103(6), 3621-3626, 1998.

[2] Maj, JB. Wouters, J. Moonen, M.: *Field Trial of a Two-Stage Adaptive Beamformer for a Dual Microphone Hearing Aid*. Ear & Hearing, (submitted).

[3] Spriet, A. Wouters, J. Moonen, M.: *Robustness Analysis of GSVD Based Optimal Filtering and Generalized Sidelobe Canceller for Hearing Aid Applications*. WASPAA, New Paltz, 21-24 October 2001.

[4] Noise Signals ICRA: *International Collegium of Rehabilitative Audiology*. Compact Disk, Ver 0.3 1997.

[5] Plomp, R. Mimpen, AM.: *Improving the Reliability of Testing the Speech Reception Threhold for Sentences*. Audiology, 18, 43-52, 1979.

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