COMBINED FEEDFORWARD-FEEDBACK NOISE REDUCTION SCHEMES FOR OPEN-FITTING HEARING AIDS

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ABSTRACT

Over the past years, the usage of open-fitting hearing aids has been steadily increasing, due to the fact that they largely alleviate occlusion-related problems. However, existing noise reduction (NR) schemes that only use the external microphones on the hearing aid do not take into account the ambient noise leaking through the open fitting, such that this leakage may even counteract the noise reduction performed by the hearing aid. Integrating active noise control (ANC) motivated algorithms into existing NR schemes by exploiting a built-in internal (error) microphone in the ear canal, enables to partially compensate for this noise leakage.

This paper presents a combined feedforward-feedback ANC (FF-FB ANC) scheme for open-fitting hearing aids. For different gains of the hearing aid, it is shown that the SNR improvement of the combined FF-FB ANC scheme outperforms a feedforward ANC and a classic Multichannel Wiener Filter noise reduction scheme without ANC. In addition, the frequency-dependent SNR improvement is analysed for different microphone positions, i.e., at the entrance of the ear canal and at the eardrum.

Index Terms— active noise control, open-fitting hearing aid, noise reduction, Multichannel Wiener Filter

1. INTRODUCTION

Understanding speech in the presence of background noise is still one of the major problems for hearing-impaired persons. Although most digital hearing aids currently make use of advanced digital signal processing and multiple microphones to achieve noise reduction, the majority of hearing aid users still reports a significant reduction in speech understanding in noisy environments. Whereas open fittings, i.e., using an ear mould with a large vent, substantially reduce the occlusion effect (caused by the ear mould – partially – sealing the ear canal from the acoustical environment) and hence improve the physical comfort [1], they have a number of disadvantages. Firstly, external sounds reach the eardrum both through the receiver of the hearing aid and directly through the vent, such that the mixing of the amplified and delayed sound produced by the hearing aid and the direct sound can have undesired perceptual effects, especially when the hearing aid processing delay is large and the hearing aid amplification is limited [2]. Secondly, the performance of existing noise reduction (NR) algorithms in the hearing aid can be severely degraded by the direct leakage of background noise through the vent. Thirdly, due to the open fitting the risk of acoustic feedback drastically increases, requiring improved acoustic feedback cancellation techniques. In this paper we will only focus on the second problem.

Current NR techniques, such as the Multichannel Wiener Filter (MWF) [3], largely ignore the occurrence of signal leakage in open-fitting hearing aids. Recent miniaturization advances however enable to incorporate an internal microphone in the ear mould, which provides information about the signal leakage and hence enables to reduce undesired perceptual effects and improve the performance of NR algorithms. Although active noise control (ANC) techniques are frequently used in (closed) headphones [4] [5], their usage in hearing aids has been quite limited. In [6] ANC has been used for reducing the occlusion effect in closed fitting hearing aids, and in [7] a feedforward ANC (FF ANC) noise reduction scheme in open-fitting hearing aids has been introduced.

In this paper, a novel combined feedforward-feedback multichannel noise reduction algorithm for open-fitting hearing aids is presented, which uses the leakage signal in the error microphone as an additional input signal to achieve speech enhancement. In addition, its performance is compared to the existing FF ANC and standard MWF algorithms.

2. SIGNAL MODEL

Considering a hearing aid with M external microphones, the mth microphone signal Y_m in the frequency-domain can be written as

$$Y_m(\omega) = X_m(\omega) + V_m(\omega), \quad m = 1 \ldots M, \tag{1}$$

with $X_m(\omega)$ the speech component and $V_m(\omega)$ the additive noise component. For conciseness the frequency-domain variable ω will be omitted in the remainder of the paper. The M-dimensional stacked vector $\mathbf{Y}$, consisting of all microphone signals, is defined as

$$\mathbf{Y} = [Y_1\ Y_2\ ...\ Y_M]^T = \mathbf{X} + \mathbf{V}. \tag{2}$$

The noise is assumed to be uncorrelated with the speech signal and the noise, speech and speech + noise correlation matrices are defined as $\mathbf{R}_v = \mathcal{E}\{\mathbf{V}\mathbf{V}^H\}$, $\mathbf{R}_x = \mathcal{E}\{\mathbf{X}\mathbf{X}^H\}$ and $\mathbf{R}_y = \mathcal{E}\{\mathbf{Y}\mathbf{Y}^H\}$, respectively.

In this paper, we are considering the usage of an ear mould with an internal (error) microphone in the ear canal (cf. Figure 1). The receiver signal Z is given by

$$Z = G\mathbf{W}^H\mathbf{Y}, \tag{3}$$

with G the (broadband) gain of the hearing aid and $\mathbf{W}$ the M-dimensional filter on the microphone signals, i.e.,

$$\mathbf{W} = [W_1\ W_2\ ...\ W_M]^T. \tag{4}$$

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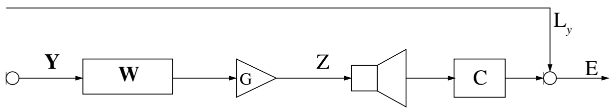

Figure 1: Hearing aid configuration with internal (error) microphone and signal leakage

The error microphone signal

$$E = CZ + L_y \tag{5}$$

consists of two terms:

- the receiver signal Z filtered with the so-called secondary path C (transfer function from the hearing aid receiver to the error microphone, including the receiver and microphone characteristics) and
- the leakage signal L_y.

Similarly we can define the signal arriving at the eardrum as

$$E^{\text{ear}} = TZ + L_y^{\text{ear}}, \tag{6}$$

with L_y^{ear} the leakage signal at the eardrum and T the so-called tertiary path (transfer function from the hearing aid receiver to the eardrum). Although these signals are typically not available for algorithmic processing, they can be recorded using an artificial head eardrum microphone.

3. MULTICHANNEL WIENER FILTER (MWF)

The goal of the multichannel Wiener filter for noise reduction is to minimize the mean-square error between the receiver signal Z and a desired signal D, i.e.,

$$J_{\text{MWF}}(\mathbf{W}) = \mathcal{E}\{|Z - D|^2\} = \mathcal{E}\{|G\mathbf{W}^H\mathbf{Y} - D|^2\}, \tag{7}$$

where D is chosen to be equal to the (unknown) speech component in a reference microphone (e.g., the first microphone), up to a delay Δ and the hearing aid gain G, i.e.,

$$D = GX_1 e^{-j\omega\Delta}. \tag{8}$$

The filter minimizing the cost function in (7) is equal to

$$\boxed{\mathbf{W}_{\text{MWF}} = \mathbf{R}_y^{-1}\mathbf{R}_x\mathbf{e}_{1,\Delta}} \tag{9}$$

with $\mathbf{e}_{1,\Delta} = [e^{j\omega\Delta} \quad \cdots \quad 0 \quad \cdots \quad 0]^T$. The receiver signal of the MWF is then equal to

$$Z_{\text{MWF}} = G\mathbf{W}_{\text{MWF}}^H\mathbf{Y}. \tag{10}$$

It should be noted that the MWF minimizing the cost function in (7) does not take into account the signal leakage L_y through the open fitting, such that the performance of the MWF will be degraded by leakage (cf. experimental results in Section 5.2).

4. INTEGRATED NOISE REDUCTION AND ACTIVE NOISE CONTROL (ANC)

In order to take into account the signal leakage L_y, we use an active ear mould, i.e., with an internal microphone in the ear canal, which records the sound signal in the ear canal, hence providing information about the signal leakage. Although at present commercial hearing aids do not have an ear canal microphone, it is possible to envisage such a microphone in the ear mould in the near future.

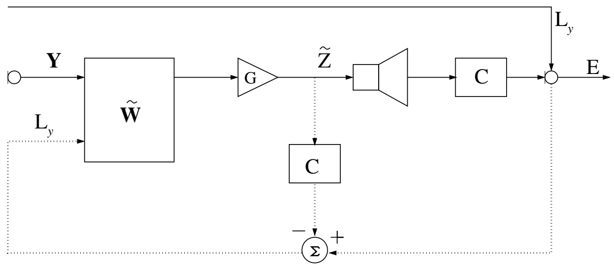

Figure 2: Combined FF-FB ANC motivated algorithm

The aim of the ANC motivated algorithms now is to minimize the difference between the signal delivered at the error microphone (including leakage) and the desired signal. Hence, on the one hand the goal is to actively suppress the noise component of the leakage signal L_y, while on the other hand the speech component should be kept with as less distortion as possible.

In the next sections we will discuss a feedforward and a combined feedforward-feedback ANC motivated algorithm for speech enhancement in open-fitting hearing aids.

4.1. Feedforward ANC (FF ANC)

In contrast to the MWF cost function in (7), where only the external microphones of the hearing aid are used, the FF ANC motivated algorithm, proposed in [7], uses both external and internal microphones, to optimize the filter $\mathbf{W}$ by minimizing the cost function

$$J_{\text{FF}}(\mathbf{W}) = \mathcal{E}\{|E - D|^2\} = \mathcal{E}\{|CZ + L_y - D|^2\}, \tag{11}$$

which now exploits information about the signal leakage L_y. The filter minimizing the cost function in (11) is then given by

$$\boxed{\mathbf{W}_{\text{FF}} = (GC^*\mathbf{R}_y)^{-1}(G\mathbf{R}_x\mathbf{e}_{1,\Delta} - \mathbf{r}_{yl_y})} \tag{12}$$

with $\mathbf{r}_{yl_y} = \mathcal{E}\{\mathbf{Y}L_y^*\}$ the cross-correlation vector between the microphones and the leakage signal. The receiver signal of the FF ANC motivated algorithm is then equal to

$$Z_{\text{FF}} = G\mathbf{W}_{\text{FF}}^H\mathbf{Y}. \tag{13}$$

4.2. Combined Feedforward-Feedback ANC (FF-FB ANC)

In this paper, we propose a combined FF-FB ANC motivated algorithm, where the leakage signal in the error microphone – unlike in the FF ANC motivated scheme – is used as an additional input signal together with the external microphones (cf. Figure 2), i.e.,

$$E_{\text{FF-FB}} = C\widetilde{Z} + L_y, \tag{14}$$

with

$$\widetilde{Z} = G\widetilde{\mathbf{W}}^H\widetilde{\mathbf{Y}} \qquad \text{and} \qquad \widetilde{\mathbf{Y}} = \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix}. \tag{15}$$

Hence, using the same desired signal (8) as in the algorithms above, the cost function for the combined FF-FB ANC motivated algorithm is given by

$$J_{\text{FF-FB}}(\widetilde{\mathbf{W}}) = \mathcal{E}\{|E_{\text{FF-FB}} - D|^2\} = \mathcal{E}\{|C\widetilde{Z} + L_y - D|^2\}. \tag{16}$$

The filter minimizing (16) is equal to

$$\boxed{\widetilde{\mathbf{W}}_{\text{FF-FB}} = (GC^*\widetilde{\mathbf{R}}_y)^{-1}(G\widetilde{\mathbf{R}}_x\mathbf{e}_{1,\Delta} - \widetilde{\mathbf{r}}_{yl_y})} \tag{17}$$

where $\widetilde{\mathbf{R}}_y = \mathcal{E}\{\widetilde{\mathbf{Y}}\widetilde{\mathbf{Y}}^H\}$, $\widetilde{\mathbf{R}}_x = \mathcal{E}\{\widetilde{\mathbf{X}}\widetilde{\mathbf{X}}^H\}$ and $\widetilde{\mathbf{r}}_{yl_y} = \mathcal{E}\{\widetilde{\mathbf{Y}}L_y^*\}$.

The receiver signal of the combined FF-FB ANC motivated algorithm is then equal to

$$Z_{\text{FF-FB}} = G\widetilde{\mathbf{W}}^H_{\text{FF-FB}}\widetilde{\mathbf{Y}}. \tag{18}$$

In order to estimate the leakage signal L_y in the error microphone, the receiver signal is filtered with the secondary path estimate and subtracted from the error signal (cf. Figure 2). Here we assume that the secondary path estimate is perfect, i.e., equal to C, which can be achieved for example by using a calibration measurement procedure.

Note that when we consider the eardrum microphone instead of the error microphone (i.e., the objective is to estimate the desired signal at the eardrum), the filters $\mathbf{W}^{\text{ear}}_{\text{FF}}$ and $\widetilde{\mathbf{W}}^{\text{ear}}_{\text{FF-FB}}$ can be computed in a similar way as in (12) and (17) by replacing E with E^{ear}, $E_{\text{FF-FB}}$ with $E^{\text{ear}}_{\text{FF-FB}}$, C with T and L_y with L_y^{ear}.

5. EXPERIMENTAL RESULTS

5.1. Setup and performance measures

Simulations were performed using anechoic room recordings obtained with a KEMAR head and torso, a two-microphone behind-the-ear (BTE) hearing aid, an external receiver (Knowles, TWFK-30017-000) and an active ear mould with an internal microphone (Knowles, FG-23329-PO7) and a vent size of 2 mm.

The sound sources were positioned at 3 m from the center of the head. The BTE was worn on the right ear. The speech source was located at 0° and multiple noise sources at 90°, 180° and 270° were considered. The noise signal was multitalker babble noise and the speech signal was composed of four sentences from the HINT database [8] ($f_s = 16$ kHz).

The used filter length is $L = 128$ and the delay Δ of the desired signal is set to half of the filter length. The first $L_c = 128$ taps of the measured secondary path C and tertiary path T have been considered.

To investigate the frequency-dependent performance of the ANC motivated algorithms, the frequency-dependent signal-to-noise ratio (SNR) improvement is computed, which is defined as the difference between the SNR of the reference microphone and the output signal (either on the error or on the KEMAR microphone)

$$\Delta\text{SNR}_j = 10\log_{10}\frac{P_{j,e_x}}{P_{j,e_v}} - 10\log_{10}\frac{P_{j,x_1}}{P_{j,v_1}}, \tag{19}$$

where j denotes frequency bin and P_{j,e_x} and P_{j,e_v} denote the power spectral density (PSD) of the speech and noise components of the output signal and P_{j,x_1} and P_{j,v_1} are similarly defined for the reference microphone signal.

In order to quantify the broadband performance of the developed noise reduction algorithms, the speech intelligibility-weighted SNR improvement [9] has been used, which takes into account the band importance function I_j, i.e.,

$$\Delta\text{SNR}_{int} = \sum_{j=1}^{J} I_j \Delta\text{SNR}_j. \tag{20}$$

The speech intelligibility-weighted input SNR in the reference microphone was 0 dB.

For the performance analysis, we define three different situations, related to which microphone has been used for computing the filter coefficients and at which (microphone) position the performance has been computed:

- case ER-ER: Both the filters $\mathbf{W}_{\text{FF}}$ in (12) , $\widetilde{\mathbf{W}}_{\text{FF-FB}}$ in (17) and the performance are calculated at the error microphone, i.e., using the signals E in (5) and $E_{\text{FF-FB}}$ in (14).
- case KE-KE: Both the filters $\mathbf{W}^{\text{ear}}_{\text{FF}}$, $\widetilde{\mathbf{W}}^{\text{ear}}_{\text{FF-FB}}$ and the performance are computed at the KEMAR microphone, i.e., using the signals E^{ear} and $E^{\text{ear}}_{\text{FF-FB}}$.
- case ER-KE: In the above mentioned cases ER-ER and KE-KE, the filter and the performance are computed at the same microphone position. On the contrary, in the case ER-KE the performance is computed at the KEMAR microphone (i.e., using the signals E^{ear}, $E^{\text{ear}}_{\text{FF-FB}}$), while the filters ($\mathbf{W}_{\text{FF}}$, $\widetilde{\mathbf{W}}_{\text{FF-FB}}$) are computed at the error microphone, in order to investigate the robustness of the filters computed at the error microphone.

5.2. Results

Figure 3 depicts ΔSNR_{int} at the error microphone for the MWF (with/without leakage, cf. Section 3) and for the ANC motivated algorithms (cf. Section 4), where the broadband gain G varies from 0 dB to 70 dB. Figure 3 shows that signal leakage degrades the performance of the MWF, especially for small G, which is particularly relevant for mild to moderate hearing impaired users who are commonly wearing open-fitting hearing aids. In addition, it can be seen from this figure that the FF ANC motivated algorithm outperforms the MWF for all gains. Since for larger values of G the ratio between the signal leakage and the receiver signal becomes smaller, the FF ANC motivated algorithm converges to the performance of the MWF without leakage [7].

For all considered gain values G, the proposed combined FF-FB ANC motivated algorithm yields the highest performance. This can be explained by the fact that in this case the leakage signal is used as an additional input signal, i.e., information from three microphone signals is being used.

For the case where the broadband gain is equal to $G = 10$ dB, we analyse the frequency-dependent ΔSNR_j. Figure 4 depicts ΔSNR_j of the FF ANC and the combined FF-FB ANC motivated algorithms for the case ER-ER. It can be observed that the performance of the combined FF-FB ANC motivated algorithm exceeds the performance of the FF ANC motivated algorithm for nearly all frequencies.

In Figure 5, ΔSNR_{int} of the FF ANC and the combined FF-FB ANC motivated algorithms is shown for the cases ER-KE and KE-KE, i.e., the performance for both cases is computed at the KEMAR microphone, but the filter estimation occurs at different microphone positions. Figure 6 compares the frequency-dependent ΔSNR_j of the combined FF-FB ANC motivated algorithm for the cases ER-KE and KE-KE.

As expected, the performance for the case KE-KE where the filter and the performance are computed at the same microphone, is better than for the case ER-KE where the filter and the performance are computed at different microphone positions. Nevertheless, for both algorithms there is only a small difference between the (broadband) performance for these two cases (cf. Figure 5). When analysing the frequency-dependent performance (cf. Figure 6), it can be observed that the difference is however most pronounced at frequencies below 400 Hz, which should be further investigated.

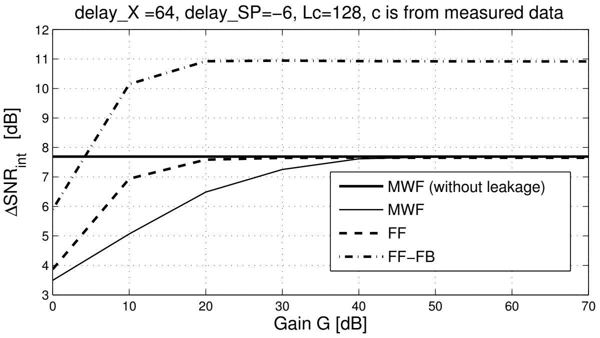

Figure 3: Speech intelligibility-weighted SNR improvement ΔSNR_{int} of the MWF (with/without leakage) and ANC motivated algorithms at the error microphone (case ER-ER).

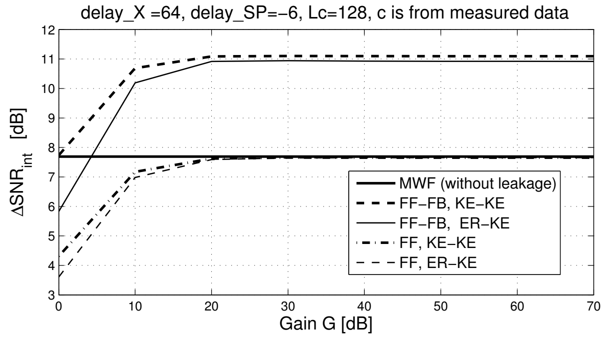

Figure 5: Speech intelligibility-weighted SNR improvement ΔSNR_{int} of the FF ANC and the combined FF-FB ANC motivated algorithms at the KEMAR microphone (cases KE-KE and ER-KE).

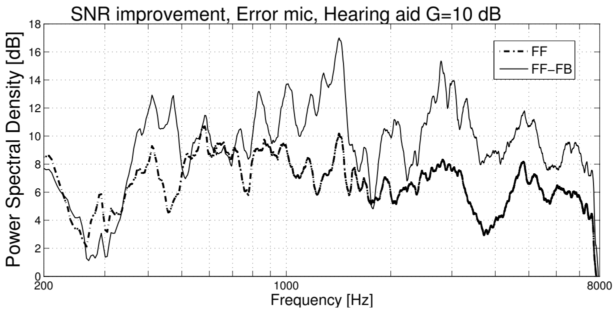

Figure 4: Frequency-dependent SNR improvement ΔSNR_j of the ANC motivated algorithms at the error microphone (case ER-ER, $G = 10$ dB).

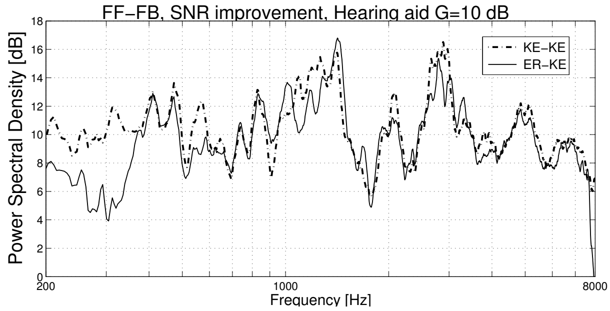

Figure 6: Frequency-dependent SNR improvement ΔSNR_j of the combined FF-FB ANC motivated algorithm, at the KEMAR microphone (cases KE-KE and ER-KE, $G = 10$ dB).

6. CONCLUSION

Present signal processing techniques for open-fitting hearing aids disregard the occurrence of signal leakage, typically leading to a degraded NR performance, especially for a small gain G.

In this paper, we have shown that using an active ear mould, i.e., with a built-in (internal) microphone, the NR performance can be improved when the internal microphone signal is used for optimizing the filter coefficients. It has been shown that the proposed combined FF-FB ANC motivated algorithm outperforms the standard MWF and FF ANC motivated algorithm for noise reduction.

Moreover, it has been shown that the performance computed at the KEMAR microphone using the filter computed at the error microphone is hardly degraded, showing the robustness of the proposed approach.

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