Integrated Active Noise Control and Noise Reduction Schemes in Open-Fitting Hearing Aids

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Introduction

Understanding speech in the presence of background noise is 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 [1], 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 and hence improve the physical comfort, current signal processing techniques in open-fitting hearing aids largely ignore the occurrence of signal leakage, which reaches the ear canal directly through the vent of the ear mould. 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 improve the performance of noise reduction algorithms. In this paper, two multichannel noise reduction algorithms, exploiting this internal microphone signal, are presented and their performance is compared.

Signal Model

We denote the mth external microphone signal of the hearing aid in the frequency-domain as

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

with X_m the desired speech component and V_m the additive noise component. 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. Furthermore, we assume that the noise correlation matrix $\mathbf{R}_v = \mathcal{E}\{\mathbf{V}\mathbf{V}^H\}$ can be estimated during noise-only periods and that the speech + noise correlation matrix $\mathbf{R}_y = \mathcal{E}\{\mathbf{Y}\mathbf{Y}^H\}$ can be estimated during speech periods, hence requiring a voice activity detector (VAD).

In this paper we are considering the usage of an ear mould with an internal (error) microphone in the ear canal. The error microphone signal can be written as

$$E = GC\mathbf{W}^H\mathbf{Y} + L_y, \tag{3}$$

with $\mathbf{W}$ the M-dimensional filter on the microphone signals, i.e.

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

and L_y the leakage signal through the vent, C the secondary path transfer function between the hearing aid receiver and the error microphone and G the (broadband) gain of the hearing aid (see Figure 1).

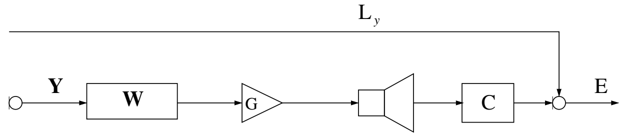

Figure 1: Hearing aid configuration with leakage

Multichannel Wiener Filter (MWF)

The goal of the considered multichannel noise reduction algorithms is to minimize the mean-square error between the output signal $Z = \mathbf{W}^H\mathbf{Y}$ and a desired signal D, i.e.

$$J_{\text{MSE}}^{\text{NR}}(\mathbf{W}) = \mathcal{E}\{|G\mathbf{W}^H\mathbf{Y} - D|^2\}, \tag{5}$$

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

$$D = GX_1 e^{-j\omega\Delta}. \tag{6}$$

It should be noted that the Multichannel Wiener filter minimising the cost function in (5) does not take into account the signal leakage L_y.

Feedforward ANC (FF ANC)

In contrast to the MWF cost function in (5), where only the external microphones of the hearing aids are used, the FF ANC motivated algorithm, proposed in [2], uses both external and internal microphones, to optimize the filter $\mathbf{W}$ by minimizing the cost function

$$\begin{aligned} J_{\text{MSE}}^{\text{FF}}(\mathbf{W}) &= \mathcal{E}\{|E - D|^2\} \\ &= \mathcal{E}\{|GC\mathbf{W}^H\mathbf{Y} + L_y - D|^2\}, \end{aligned} \tag{7}$$

which now exploits information about the signal leakage L_y.

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 is - unlike in the FF ANC motivated scheme - used as an additional input signal together with the external microphones (see Figure 2), i.e.

$$E^{\text{FF-FB}} = GC\mathbf{W}^H \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix} + L_y. \tag{8}$$

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

$$\begin{aligned} J_{\text{MSE}}^{\text{FF-FB}}(\mathbf{W}) &= \mathcal{E}\{|E^{\text{FF-FB}} - D|^2\} \\ &= \mathcal{E}\{|GC\mathbf{W}^H \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix} + L_y - D|^2\}. \end{aligned} \tag{9}$$

In order to estimate the leakage signal L_y in the error microphone, the receiver signal is filtered with the estimated secondary path $\widehat{C}$ (which we assume here to be equal to the secondary path C) and is subtracted from the error signal.

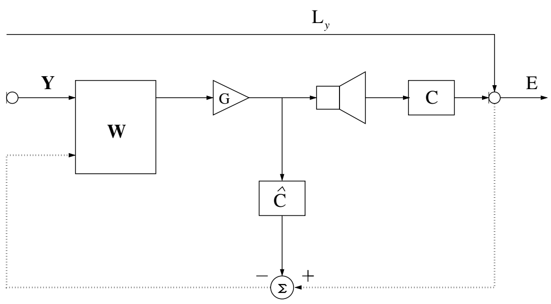

Figure 2: Combined FF-FB ANC motivated algorithm

Experimental Results

Simulations were performed on anechoic room recordings obtained with a KEMAR head and torso, using 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).

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

In order to quantify the performance of the developed noise reduction algorithms, the speech intelligibility-weighted signal-to-noise ratio (SNR) [3] has been used, which takes into account the band importance function I_j and the SNR of the jth band (SNR_j), i.e.

$$\text{SNR}_{int} = \sum_{j=1}^{J} I_j SNR_j \tag{10}$$

The input SNR was 0dB in the reference microphone.

Figure 3 depicts the intelligibility-weighted output SNR calculated on the error microphone for the MWF (with/without leakage) and the ANC motivated algorithms, where the gain G varies from 0dB to 70dB. The delay Δ of the desired signal is set to half of the filter length ($L = 128$). The first $L_c = 128$ taps of the measured secondary path C have been considered.

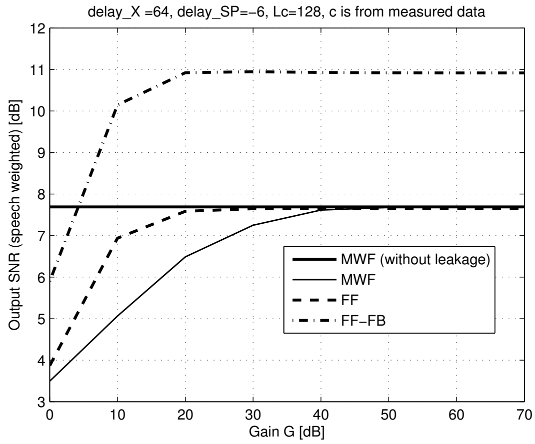

Figure 3: Performance comparison for MWF (with/without leakage) and ANC motivated algorithms

Figure 3 shows that signal leakage seriously degrades the performance of the MWF, especially for small G. The FF ANC motivated algorithm outperforms the MWF for all G. When G is large, the performance of the FF ANC motivated algorithm converges to the performance of the MWF without leakage.

For all values of G, the best performance is obtained by the proposed combined FF-FB ANC motivated algorithm, which can be explained by the fact that in this case the leakage signal is used as an additional input signal.

Conclusion

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

References

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