Effect of Secondary Path Amplitude Estimation Errors on the Performance of ANC-Motivated Algorithms for Open-Fitting Hearing Aids

Derya Dalga, Simon Doclo
University of Oldenburg, Department of Medical Physics and Acoustics - Signal Processing Group
Email: {derya.dalga, simon.doclo}@uni-oldenburg.de

Introduction

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 such as the multichannel Wiener filter (MWF) [1] do not take into account the ambient noise leaking through the open fitting, therefore typically leading to a degraded noise reduction performance. To provide information about this leakage and hence improve the performance of NR algorithms, an internal (so-called error) microphone can be incorporated in the ear mould.

In [2] a feedforward active noise control-motivated (FF ANC) algorithm in open-fitting hearing aids has been introduced, which takes into account the signal leakage for the filter optimization. In [3], a combined feedforward-feedback active noise control-motivated (FF-FB ANC) algorithm for open-fitting hearing aids has been presented, which uses the signal leakage in the error microphone as an additional input signal together with the external microphone signals. It has been experimentally shown that the FF-FB ANC algorithm outperforms both the standard MWF and the FF ANC algorithms in terms of SNR improvement. In [4], a theoretical analysis has been performed for the standard MWF and the ANC-motivated algorithms. It has been theoretically shown that if a perfect estimate of the so-called secondary path (i.e., the transfer function from the hearing aid receiver to the error microphone) is available, the error microphone signal of the FF-FB ANC algorithm is independent of the signal leakage.

Since in practice the secondary path is typically unknown and can change over time, it needs to be estimated. The performance of the ANC-motivated algorithms is affected by errors in the secondary path estimation. Therefore, in this paper we analyse the influence of secondary path amplitude estimation errors on the performance of the ANC-motivated algorithms. We will show that a closed-form expression for the filter of the FF ANC algorithm can be derived as a function of the secondary path estimation error. Moreover, we will show that even when amplitude estimation errors occur the FF-FB ANC algorithm still outperforms the FF ANC algorithm.

This work was supported in part by the PhD Program "Function and Pathophysiology of the Auditory System", funded by the state of Lower Saxony, and the Research Unit FOR 1732 "Individualized Hearing Acoustics", funded by the German Research Foundation (DFG).

Configuration and Notation

Consider a hearing aid with M external microphones and an internal (error) microphone in the ear canal, as depicted in Figure 1. The mth microphone signal $Y_m(k,n)$ in the short-time Fourier transform domain can be written as

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

with $X_m(k,n)$ the speech component and $V_m(k,n)$ the additive noise component, where k denotes the frequency index and n the block index. For conciseness the indices k and n will be omitted in the remainder of the paper. The M-dimensional stacked vector $\mathbf{Y}$, consisting of all microphone signals, is defined as

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

The correlation matrices of the signal components are defined as $\mathbf{R}_v = \mathcal{E}\{\mathbf{V}\mathbf{V}^H\}$, $\mathbf{R}_x = \mathcal{E}\{\mathbf{X}\mathbf{X}^H\}$ and $\mathbf{R}_y = \mathcal{E}\{\mathbf{Y}\mathbf{Y}^H\}$. The error microphone signal E is equal to

$$E = CZ + L_y, \tag{3}$$

with C the secondary path and L_y the signal leakage through the open fitting. The receiver signal Z is given by

$$Z = G\mathbf{W}^H\mathbf{Y}, \tag{4}$$

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

$$\mathbf{W} = [W_1\ W_2\ \ldots\ W_M]^T. \tag{5}$$

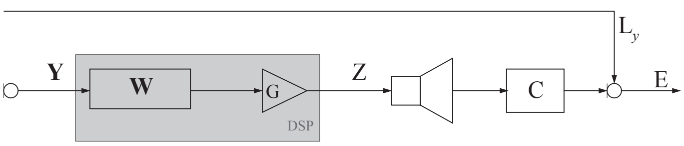

Figure 1: Hearing aid configuration with external microphones $\mathbf{Y}$, internal (error) microphone E and signal leakage L_y.

Multichannel Wiener Filter (MWF)

The multichannel Wiener filter produces a minimum-mean-square-error (MMSE) estimate of the (unknown) speech component in a reference microphone (e.g., the first microphone). The MSE cost function is given by

$$J_{\text{MSE}}^{\text{no-leakage}}(\mathbf{W}) = \mathcal{E}\{|CZ - D|^2\} = \mathcal{E}\{|GC\mathbf{W}^H\mathbf{Y} - D|^2\}, \tag{6}$$

where the desired signal D is chosen to be equal to the speech component in the first microphone, multiplied with the hearing aid gain and filtered with the secondary path, i.e.,

$$D = GCX_1. \tag{7}$$

The filter minimizing the cost function in (6) is equal to

$$\mathbf{W}_{\text{MWF}} = \mathbf{R}_y^{-1}\mathbf{R}_x\mathbf{e}_1, \tag{8}$$

with $\mathbf{e}_i$ a vector whose ith element is equal to 1 and all other elements are equal to 0. Note that this filter 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 this signal leakage [3].

Effect of Secondary Path Estimation Errors

In order to take into account the signal leakage, the feedforward and combined feedforward-feedback ANC-motivated algorithms proposed in [2] and [3] use both the external microphone signals and the error microphone signal E, providing information about the signal leakage L_y. The aim is then to minimize the MSE between the error microphone signal E (including leakage) and the desired signal D.

If a perfect estimate of the secondary path is available (e. g. using a calibration measurement procedure), then a closed-form expression for the FF ANC and FF-FB ANC filters can be derived [3].

In practice the secondary path is typically estimated , e.g. using an adaptive filtering algorithm as in [5], such that estimation errors may occur. The estimated secondary path $\widehat{C}$ can be written as

$$\boxed{\widehat{C} = CA}, \tag{9}$$

where A represents the secondary path estimation error. In this paper, we will analyse the effect of secondary path amplitude estimation errors for the FF ANC and FF-FB ANC algorithms, both analytically as well as using simulations.

Feedforward ANC-Motivated (FF ANC) Algorithm

In contrast to the MSE cost function in (6), the FF ANC algorithm [2], minimizes the cost function

$$J_{\text{MSE}}^{\text{leakage}}(\mathbf{W}) = \mathcal{E}\{|E-D|^2\} = \mathcal{E}\{|CZ + L_y - D|^2\}, \tag{10}$$

The filter minimizing the cost function in (10) is then given by

$$\mathbf{W}_{\text{FF}}^{\text{opt}} = (GC^*\mathbf{R}_y)^{-1}(GC^*\mathbf{R}_x\mathbf{e}_1 - \mathbf{r}_{yl_y}), \tag{11}$$

with $\mathbf{r}_{yl_y} = \mathcal{E}\{\mathbf{Y}L_y^*\}$. The filter in (11) can be related to the MWF in (8) as

$$\mathbf{W}_{\text{FF}}^{\text{opt}} = \mathbf{W}_{\text{MWF}} - (GC^*)^{-1}\mathbf{R}_y^{-1}\mathbf{r}_{yl_y}. \tag{12}$$

In practice we do not have the perfect estimate of the secondary path and the signal leakage. The filter of the FF ANC algorithm with the estimated secondary path and the estimated signal leakage can be expressed as

$$\mathbf{W}_{\text{FF}} = \mathbf{W}_{\text{MWF}} - (G\widehat{C}^*)^{-1}\mathbf{R}_y^{-1}\mathbf{r}_{y\hat{l}_y}, \tag{13}$$

with $\mathbf{r}_{y\hat{l}_y} = \mathcal{E}\{\mathbf{Y}\widehat{L}_y^*\}$. In order to estimate the signal leakage in the error microphone, the receiver signal is filtered with the secondary path estimate and subtracted from the error signal, i.e.,

$$\widehat{L}_y = E - \widehat{C}Z = G(C-\widehat{C})\mathbf{W}_{\text{FF}}^H\mathbf{Y} + L_y. \tag{14}$$

Note that the estimated signal leakage $\widehat{L}_y$ depends on the filter $\mathbf{W}_{\text{FF}}$. Using (14), the cross correlation vector $\mathbf{r}_{y\hat{l}_y}$ can be written as

$$\mathbf{r}_{y\hat{l}_y} = \mathcal{E}\{\mathbf{Y}\widehat{L}_y^*\} = G(C-\widehat{C})^*\mathbf{R}_y\mathbf{W}_{\text{FF}} + \mathbf{r}_{yl_y}. \tag{15}$$

Inserting (15) in (13) yields a closed-form expression for the filter $\mathbf{W}_{\text{FF}}$, i.e.,

$$\boxed{\mathbf{W}_{\text{FF}} = \mathbf{W}_{\text{MWF}}\left(\frac{\widehat{C}}{C}\right)^* - (GC^*)^{-1}\mathbf{R}_y^{-1}\mathbf{r}_{yl_y}} \tag{16}$$

Note that the filter in (16) is independent of the estimated signal leakage $\widehat{L}_y$ and only the first part is dependent on the secondary path estimation error.

Combined Feedforward-Feedback ANC-Motivated (FF-FB ANC) Algorithm

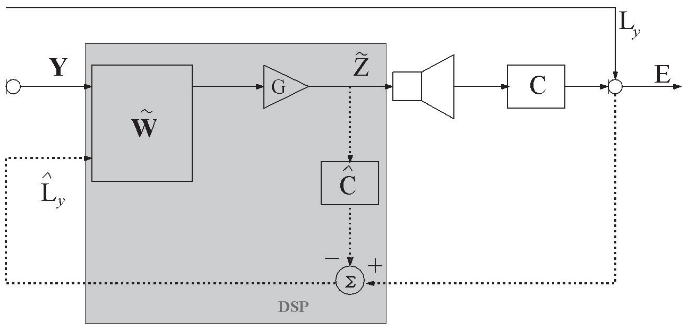

Figure 2: FF-FB ANC algorithm scheme

In the FF-FB ANC algorithm proposed in [3], the signal leakage in the error microphone – unlike in the FF ANC algorithm – is used as an additional input signal together with the external microphones (cf. Figure 2), i.e.,

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

with

$$\widetilde{Z} = G\widetilde{\mathbf{W}}^H\widetilde{\mathbf{Y}} \qquad \text{and} \qquad \widetilde{\mathbf{Y}} = \begin{bmatrix}\mathbf{Y}\\ \widehat{L}_y\end{bmatrix}. \tag{17}$$

The cost function for the FF-FB ANC algorithm is given by

$$J_{\text{MSE}}^{\text{leakage}}(\widetilde{\mathbf{W}}) = \mathcal{E}\{|E_{\text{FF-FB}} - D|^2\} = \mathcal{E}\{|C\widetilde{Z} + L_y - D|^2\}. \tag{18}$$

The filter minimizing the cost function in (18) is equal to

$$\widetilde{\mathbf{W}}_{\text{FF-FB}}^{\text{opt}} = (GC^*\widetilde{\mathbf{R}}_y^{\text{opt}})^{-1}(GC^*\widetilde{\mathbf{R}}_x^{\text{opt}}\mathbf{e}_1 - \widetilde{\mathbf{r}}_{yl_y}^{\text{opt}}), \tag{19}$$

with $\widetilde{\mathbf{r}}^{\text{opt}}_{yl_y} = \mathcal{E}\{\widetilde{\mathbf{Y}}^{\text{opt}} L_y^*\}$ and $\widetilde{\mathbf{R}}^{\text{opt}}_y$, $\widetilde{\mathbf{R}}^{\text{opt}}_x$ the correlation matrices of the microphone signals $\widetilde{\mathbf{Y}}^{\text{opt}} = \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix}$ and their speech components, respectively.

In practice we do not have the perfect estimate of the secondary path and the signal leakage. The filter of the FF-FB ANC algorithm with the estimated secondary path and the estimated signal leakage can be expressed as

$$\widetilde{\mathbf{W}}_{\text{FF-FB}} = (G\widehat{C}^*\widetilde{\mathbf{R}}_y)^{-1}(G\widehat{C}^*\widetilde{\mathbf{R}}_x\mathbf{e}_1 - \widetilde{\mathbf{r}}_{y\hat{l}_y}). \tag{20}$$

Using the fact that $\widehat{L}_y = \mathbf{e}^H_{\text{M+1}}\widetilde{\mathbf{Y}}$, it follows that

$$\widetilde{\mathbf{r}}_{y\hat{l}_y} = \mathcal{E}\{\widetilde{\mathbf{Y}}\widehat{L}_y^*\} = \widetilde{\mathbf{R}}_y\mathbf{e}_{\text{M+1}}, \tag{21}$$

with $\widetilde{\mathbf{R}}_y = \mathcal{E}\{\widetilde{\mathbf{Y}}\widetilde{\mathbf{Y}}^H\}$. Then the filter $\widetilde{\mathbf{W}}_{\text{FF-FB}}$ in (20) can be written as

$$\widetilde{\mathbf{W}}_{\text{FF-FB}} = \widetilde{\mathbf{W}}_{\text{MWF}} - (G\widehat{C}^*)^{-1}\mathbf{e}_{\text{M+1}}, \tag{22}$$

with $\widetilde{\mathbf{W}}_{\text{MWF}} = \widetilde{\mathbf{R}}_y^{-1}\widetilde{\mathbf{R}}_x\mathbf{e}_1$ and $\widetilde{\mathbf{R}}_x = \mathcal{E}\{\widetilde{\mathbf{X}}\widetilde{\mathbf{X}}^H\}$. Note that the first part $\widetilde{\mathbf{W}}_{\text{MWF}}$ depends on the estimated signal leakage $\widehat{L}_y$ and can be interpreted as the MWF with $\widehat{L}_y$ as an additional input signal. The second part of the filter is a vector where only the last element is not equal to zero and only depends on the gain and the estimated secondary path (i.e., independent of the estimated signal leakage $\widehat{L}_y$).

For estimating the signal leakage in the error microphone, the receiver signal is filtered with the secondary path estimate and subtracted from the error signal (cf. (14)), i.e.,

$$\widehat{L}_y = E_{\text{FF-FB}} - \widehat{C}\widetilde{Z} = G(C - \widehat{C})\widetilde{\mathbf{W}}^H_{\text{FF-FB}}\widetilde{\mathbf{Y}} + L_y. \tag{23}$$

However, by plugging (23) into (22), it is unfortunately not possible to derive a closed-form expression for the filter $\widetilde{\mathbf{W}}_{\text{FF-FB}}$ as a function of the secondary path estimation error.

Nevertheless, the filter $\widetilde{\mathbf{W}}_{\text{FF-FB}}$ can be computed iteratively as following. Starting from an initial value for the estimated signal leakage, e.g. $\widehat{L}_y = L_y$, $\widetilde{\mathbf{W}}_{\text{FF-FB}}$ can be computed using (22). By plugging the computed $\widetilde{\mathbf{W}}_{\text{FF-FB}}$ into (23) a new estimated signal leakage can be found. These iterations are then performed until the filter converges.

Experimental Results

In this section, we perform simulations to investigate the broadband performance of the ANC-motivated noise reduction algorithms, when secondary path amplitude estimation errors occur.

Experimental Setup

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 a distance of 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 taken from the HINT database [6].

The signals were processed at $f_s = 16$ kHz using a weighted overlap-add (WOLA) framework with a block size of 256 samples, an overlap of 75% between successive blocks and a Hann window. The correlation matrices $\mathbf{R}_y$, $\mathbf{R}_v$ and $\mathbf{R}_x$ are estimated as

$$\mathbf{R}_y(k) = \frac{1}{N_y}\sum_{i=1}^{N_y}\mathbf{Y}(k,i)\mathbf{Y}^H(k,i) \quad \text{speech present} \tag{24}$$

$$\mathbf{R}_x(k) = \frac{1}{N_y}\sum_{i=1}^{N_y}\mathbf{X}(k,i)\mathbf{X}^H(k,i) \quad \text{speech present} \tag{25}$$

$$\mathbf{R}_v(k) = \frac{1}{N_v}\sum_{i=1}^{N_v}\mathbf{V}(k,i)\mathbf{V}^H(k,i) \quad \text{speech absent} \tag{26}$$

with N_y the number of available signal blocks when speech is present and N_v the number of available signal blocks when speech is absent, determined by a voice activity detector (VAD).

Using a calibration measurement procedure, the (perfect) secondary path C has been measured (FIR filter with $L_c = 128$ taps). The broadband gain is set to $G = 0$ dB.

Performance Measures

To quantify the performance of the considered noise reduction algorithms, the (broadband) speech intelligibility-weighted SNR improvement [7] is computed, i.e.,

$$\Delta\text{SNR}_{int} = \sum_{k=1}^{K} I_k\Delta\text{SNR}_k, \tag{27}$$

where the weighting function I_k takes the importance of the k-th frequency band for speech intelligibility into account and

$$\Delta\text{SNR}_k = 10\log_{10}\frac{P_{k,e_x}}{P_{k,e_v}} - 10\log_{10}\frac{P_{k,x_1}}{P_{k,v_1}}, \tag{28}$$

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

In order to evaluate the amount of speech distortion, we use the intelligibility-weighted spectral distortion measure SD_{int}:

$$\text{SD}_{int} = \sum_{k=1}^{K} I_k\text{SD}_k, \tag{29}$$

with

$$\mathrm{SD}_k = 10\log_{10}\frac{P_{k,e_x}}{P_{k,d}}, \tag{30}$$

where $P_{k,d}$ denote the PSD of the desired signal D in (7).

Results

Figure 3 and 4 depict $\Delta\mathrm{SNR}_{int}$ and SD_{int} for the FF-FB ANC and FF ANC algorithms, as a function of the broadband amplitude error in the secondary path estimate. In [4] it had been already shown that for a perfect estimate of the secondary path (i.e., $A = 0$ dB) the FF-FB ANC algorithm provides a better performance than the FF ANC algorithm. Figure 3 shows that for secondary path amplitude estimation errors the SNR improvement of the FF-FB ANC algorithm is hardly degraded. Moreover, even when amplitude estimation errors occur the FF-FB ANC algorithm still outperforms the FF ANC algorithm. From Figure 4 it can be observed that the FF-FB ANC algorithm causes less speech distortion than the FF ANC algorithm nearly for all considered amplitude errors.

Conclusion

In this paper, we have derived an analytical expression for the filter of the FF ANC algorithm to analyse the effect of the secondary path estimation errors. Moreover, we have shown that even when secondary path amplitude estimation errors occur the FF-FB ANC algorithm still outperforms the FF ANC algorithm.

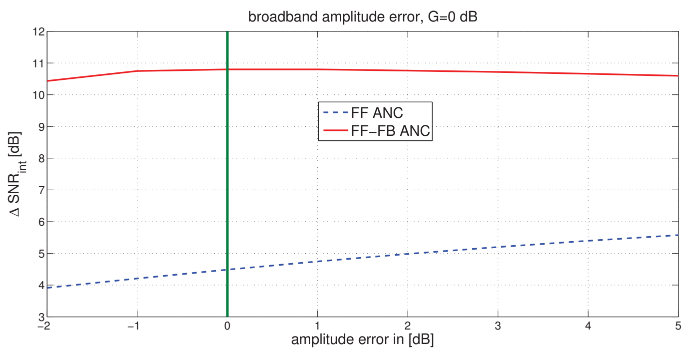

Figure 3: Speech intelligibility-weighted SNR improvement of the ANC-motivated algorithms depending on the broadband amplitude error in the secondary path estimation.

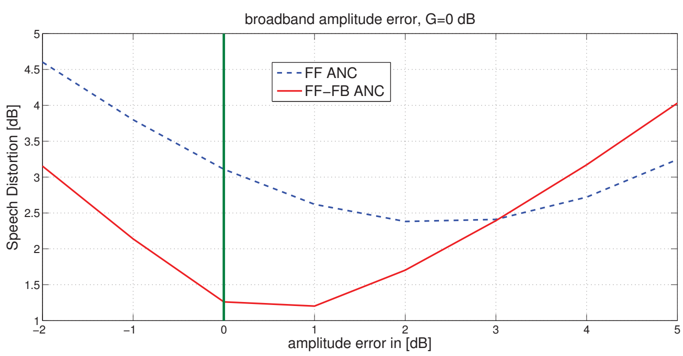

Figure 4: Speech intelligibility-weighted speech distortion of the ANC-motivated algorithms depending on the broadband amplitude error in the secondary path estimation.

References

[1] S. Doclo, A. Spriet, J. Wouters, and M. Moonen, “Frequency-domain criterion for the speech distortion weighted multichannel Wiener filter for robust noise reduction,” *Speech Communication*, vol. 49, no. 7-8, pp. 636-656, Jul.-Aug. 2007.

[2] R. Serizel, M. Moonen, J. Wouters, and S. H. Jensen, “Integrated active noise control and noise reduction in hearing aids,” *IEEE Trans. on Audio, Speech and Language Processing*, vol. 18, no. 6, pp. 1137-1146, Aug. 2010.

[3] D. Dalga, S. Doclo, “Combined feedforward-feedback noise reduction schemes for open-fitting hearing aids,” in *Proc. WASPAA*, pp. 185-188, New Paltz NY, USA, Oct. 2011.

[4] D. Dalga, S. Doclo, “Theoretical performance analysis of ANC-motivated noise reduction algorithms for open-fitting hearing aids,” in *Proc. IWAENC*, pp. 77-80, Aachen, Germany, Sep. 2012.

[5] S. M. Kuo, D. R. Morgan, “Active noise control: a tutorial review,” in *Proc. IEEE*, vol. 87, no. 6, pp. 943–973, Jun. 1999.

[6] M. Nilsson, S. D. Soli, and A. Sullivan, “Development of the hearing in noise test for the measurement of speech reception thresholds in quiet and in noise,” *Journal of the Acoustical Society of America*, vol. 95, no. 2, pp. 1085-1099, Feb. 1994.

[7] J. E. Greenberg, P. M. Peterson, and P. M. Zurek, “Intelligibility-weighted measures of speech-to-interference ratio and speech system performance,” *Journal of the Acoustical Society of America*, vol. 94, no. 5, pp. 3009-3010, Nov. 1993.