THEORETICAL PERFORMANCE ANALYSIS OF ANC-MOTIVATED NOISE REDUCTION ALGORITHMS FOR OPEN-FITTING HEARING AIDS

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ABSTRACT

Existing noise reduction techniques for open-fitting hearing aids typically disregard the occurrence of signal leakage through the open-fitting, leading to a degraded noise reduction performance. Recent miniaturization advances enable to incorporate an internal microphone in the ear mould, which is able to record this signal leakage.

Recently, feedforward and combined feedforward-feedback active-noise-control-motivated algorithms for noise reduction have been proposed for open-fitting hearing aids. In this paper, a theoretical analysis of these algorithms is presented and the output signal-to-noise ratios (SNR) for a single speech source scenario are derived. It is shown that the performance of the combined feedforward-feedback ANC-motivated (FF-FB ANC) algorithm is independent of the signal leakage. In addition the FF-FB ANC algorithm delivers the highest output SNR of all considered noise reduction algorithms.

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

1. 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. Current noise reduction (NR) techniques such as the Multichannel Wiener Filter (MWF) [1], do not take into account the ambient noise leaking through the open-fitting, leading to a degraded noise reduction performance. To provide information about this signal leakage and hence improve the performance of NR algorithms, an internal (error) microphone can be incorporated in the ear mould. Although active noise control (ANC) has been frequently used in (closed) headphones [2][3], its usage in hearing aids has been rather limited. In [4], ANC has been used for reducing the occlusion effect in closed fitting hearing aids, and in [5] a feedforward ANC-motivated (FF ANC) algorithm in open-fitting hearing aids has been introduced.

In [6], a combined feedforward-feedback ANC-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 to achieve speech enhancement. It has been shown that in terms of SNR improvement the FF-FB ANC algorithm outperforms the standard MWF and the FF ANC algorithms.

In this paper, a theoretical analysis is performed for the standard MWF and the ANC-motivated algorithms. The output SNRs for a single speech source are derived for these algorithms. The error microphone signals and the filters of the considered noise reduction algorithms are analysed for extreme cases of large and small gains. It is shown that for large hearing aid gains the standard MWF and the FF ANC algorithms achieve the same performance. Moreover, the performance of the FF-FB ANC algorithm is – remarkably – independent of the signal leakage and it has the highest output SNR of all considered algorithms.

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2. SIGNAL MODEL

Considering a hearing aid with M external microphones and an internal (error) microphone in the ear canal (cf. Figure 1), the mth microphone signal $Y_m(k,n)$ in the frequency-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 so-called secondary path (transfer function from the hearing aid receiver to the error microphone, including the receiver and microphone characteristics) 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.,

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

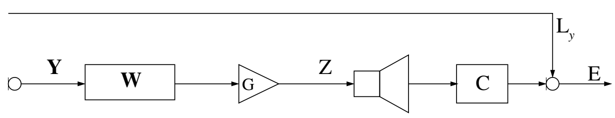

Fig. 1. Hearing aid configuration with external microphones $\mathbf{Y}$, internal (error) microphone E and signal leakage L_y.

3. 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 hence given by

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

where 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.,

$$D = GCX_1. \quad (7)$$

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

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

with $\mathbf{e}_i$ a vector whose ith element equals 1 and all other elements equal 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.

4. ACTIVE NOISE CONTROL (ANC)-MOTIVATED NOISE REDUCTION ALGORITHMS

In order to take into account the signal leakage the ANC-motivated algorithms proposed in [5] and [6] use both the external microphone signals and the error microphone signal E, providing information about the signal leakage L_y.

The aim now is to minimize the MSE between the error microphone signal E (including leakage) and the desired signal D.

4.1. FF ANC algorithm

In contrast to the MSE cost function in (6), the FF ANC algorithm [5], minimizes the cost function

$$J_{\text{MSE}}^{\text{leakage}}(\mathbf{W}) = \mathcal{E}\{|E - D|^2\} = \mathcal{E}\{|CZ + L_y - D|^2\}, \quad (9)$$

which now exploits information about the signal leakage L_y. The filter minimizing the cost function in (9) is then given by

$$\mathbf{W}_{\text{FF}} = (GC^*\mathbf{R}_y)^{-1}(GC^*\mathbf{R}_x\mathbf{e}_1 - \mathbf{r}_{yl_y}), \quad (10)$$

with $\mathbf{r}_{yl_y} = \mathcal{E}\{\mathbf{Y}L_y^*\}$. The filter in (10) can be related to the MWF in (8) as

$$\boxed{\mathbf{W}_{\text{FF}} = \mathbf{W}_{\text{MWF}} - (GC^*)^{-1}\mathbf{R}_y^{-1}\mathbf{r}_{yl_y}}. \quad (11)$$

The error microphone signal of the FF ANC algorithm is equal to

$$\boxed{E_{\text{FF}} = GC\mathbf{W}_{\text{MWF}}^H\mathbf{Y} - \mathbf{r}_{yl_y}^H\mathbf{R}_y^{-1}\mathbf{Y} + L_y}. \quad (12)$$

We will now analyse the filter in (10) and the error microphone signal in (12) for extreme cases of large and small gains. For large gain values, $\mathbf{W}_{\text{FF}}$ in (10) and E_{FF} in (12) reduce to

$$\lim_{G\to\infty} \mathbf{W}_{\text{FF}} = \mathbf{W}_{\text{MWF}}, \quad (13)$$

$$\lim_{G\to\infty} E_{\text{FF}} = GC\mathbf{W}_{\text{MWF}}^H\mathbf{Y}, \quad (14)$$

i.e., to the MWF solution without taking into account the signal leakage, which is logical since a large gain corresponds to negligible signal leakage.

For small gains[1], $\mathbf{W}_{\text{FF}}$ in (10) and E_{FF} in (12) reduce to

$$\lim_{G\to 0} \mathbf{W}_{\text{FF}} = -(GC^*)^{-1}\mathbf{R}_y^{-1}\mathbf{r}_{yl_y}, \quad (15)$$

$$\lim_{G\to 0} E_{\text{FF}} = -(\mathbf{r}_{yl_y}^H\mathbf{R}_y^{-1})\mathbf{Y} + L_y. \quad (16)$$

In this case, the MWF part in $\mathbf{W}_{\text{FF}}$ is canceled out and E_{FF} is independent of the hearing aid gain G. This can be interpreted as the standard ANC case, where the aim is to suppress the complete signal leakage, i.e., $D = 0$ and hence

$$J_{\text{ANC}}(\mathbf{W}) = \mathcal{E}\{|GC\mathbf{W}^H\mathbf{Y} + L_y|^2\}. \quad (17)$$

4.2. FF-FB ANC algorithm

In the FF-FB ANC algorithm proposed in [6], 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

$$\widetilde{Z} = G\widetilde{\mathbf{W}}^H\widetilde{\mathbf{Y}} \qquad \text{and} \qquad \widetilde{\mathbf{Y}} = \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix}. \quad (18)$$

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

$$J_{\text{MSE}}^{\text{leakage}}(\widetilde{\mathbf{W}}) = \mathcal{E}\{|E_{\text{FF-FB}} - D|^2\} = \mathcal{E}\{|C\widetilde{Z} + L_y - D|^2\}. \quad (19)$$

The filter minimizing the cost function in (19) is equal to

$$\widetilde{\mathbf{W}}_{\text{FF-FB}} = (GC^*\widetilde{\mathbf{R}}_y)^{-1}(GC^*\widetilde{\mathbf{R}}_x\mathbf{e}_1 - \widetilde{\mathbf{r}}_{yl_y}). \quad (20)$$

Using the fact that $L_y = \mathbf{e}_{\text{M+1}}^H\widetilde{\mathbf{Y}}$, it follows that

$$\widetilde{\mathbf{r}}_{yl_y} = \mathcal{E}\{\widetilde{\mathbf{Y}}L_y^*\} = \widetilde{\mathbf{R}}_y\mathbf{e}_{\text{M+1}}, \quad (21)$$

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

$$\boxed{\widetilde{\mathbf{W}}_{\text{FF-FB}} = \widetilde{\mathbf{W}}_{\text{MWF}} - (GC^*)^{-1}\mathbf{e}_{\text{M+1}}}, \quad (22)$$

with $\widetilde{\mathbf{W}}_{\text{MWF}} = \widetilde{\mathbf{R}}_y^{-1}\widetilde{\mathbf{R}}_x\mathbf{e}_1$ and $\widetilde{\mathbf{R}}_x = \mathcal{E}\{\widetilde{\mathbf{X}}\widetilde{\mathbf{X}}^H\}$.

The first part $\widetilde{\mathbf{W}}_{\text{MWF}}$, which is independent of the gain G, can be interpreted as the MWF with the signal leakage L_y as an additional input signal. The second part is a vector where only the last element is not equal to zero and only depends on the gain and the secondary path (i.e., independent of the signal leakage L_y).

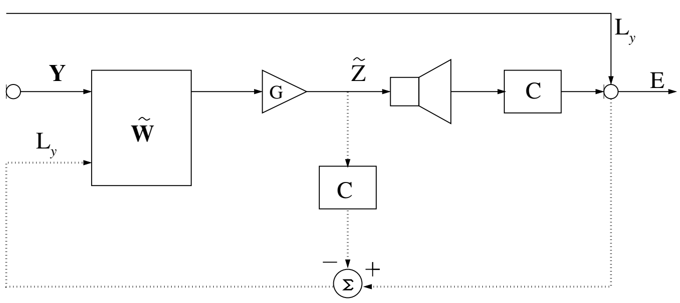

Fig. 2. FF-FB ANC algorithm scheme

[1] Small gain values are unrealistic, but they are still covered in this paper.

The error microphone signal of the FF-FB ANC algorithm is then given by

$$E_{\text{FF-FB}} = GC(\widetilde{\mathbf{W}}_{\text{MWF}} - (GC^*)^{-1}\mathbf{e}_{\text{M+1}})^H\widetilde{\mathbf{Y}} + L_y, \quad (23)$$

which reduces to

$$\boxed{E_{\text{FF-FB}} = GC\widetilde{\mathbf{W}}_{\text{MWF}}^H\widetilde{\mathbf{Y}}}. \quad (24)$$

Remarkably, the error microphone signal $E_{\text{FF-FB}}$ is independent of the signal leakage L_y and can be interpreted as the output signal of the MWF, which uses the external and internal microphones as input signals.

In practice, the performance of the FF-FB ANC algorithm depends on the signal leakage estimation error. To estimate the signal leakage 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). The secondary path estimate is considered to be perfect i.e., equal to C, which can be achieved for example using a calibration measurement procedure.

5. THEORETICAL PERFORMANCE ANALYSIS

In this section we analyse in each frequency band the output SNR of the MWF (with/without leakage) and of the ANC-motivated algorithms for a single speech source.

Assuming that a single speech source is present, the speech signal vector is given by $\mathbf{X} = \mathbf{H}S$, with $\mathbf{H}$ the M-dimensional steering vector containing the acoustic transfer functions between the speech source and the hearing aid microphones (including microphone characteristics, room acoustics and head shadow effect) and S the speech signal. Hence, the speech correlation matrix is a rank-1 matrix, i.e.,

$$\mathbf{R}_x = P_s\mathbf{H}\mathbf{H}^H, \quad (25)$$

where $P_s = \mathcal{E}\{|S|^2\}$ denotes the power of the speech signal. Using (25) and the matrix inversion lemma, the inverse matrix $\mathbf{R}_y^{-1}$ can be expressed as

$$\mathbf{R}_y^{-1} = (P_s\mathbf{H}\mathbf{H}^H + \mathbf{R}_v)^{-1} = \mathbf{R}_v^{-1} - \frac{\mathbf{R}_v^{-1}P_s\mathbf{H}\mathbf{H}^H\mathbf{R}_v^{-1}}{1+\rho}, \quad (26)$$

with

$$\rho = P_s\mathbf{H}^H\mathbf{R}_v^{-1}\mathbf{H}. \quad (27)$$

5.1. Multichannel Wiener Filter

Inserting (25) and (26) into (8) yields the well-known expresion for the MWF in the case of a single speech source, i.e.,

$$\mathbf{W}_{\text{MWF}} = P_s\frac{\mathbf{R}_v^{-1}\mathbf{H}}{1+\rho}H_1^*. \quad (28)$$

Assuming no signal leakage, the output SNR of the MWF is given by

$$SNR_{\text{MWF}}^{\text{no-leakage}} = \frac{\mathcal{E}\{|GC\mathbf{W}_{\text{MWF}}^H\mathbf{X}|^2\}}{\mathcal{E}\{|GC\mathbf{W}_{\text{MWF}}^H\mathbf{V}|^2\}} = \frac{\mathbf{W}_{\text{MWF}}^H\mathbf{R}_x\mathbf{W}_{\text{MWF}}}{\mathbf{W}_{\text{MWF}}^H\mathbf{R}_v\mathbf{W}_{\text{MWF}}} = \rho. \quad (29)$$

Taking the signal leakage into account, the output SNR is equal to

$$SNR_{\text{MWF}}^{\text{leakage}} = \frac{\mathcal{E}\{|GC\mathbf{W}_{\text{MWF}}^H\mathbf{X} + L_x|^2\}}{\mathcal{E}\{|GC\mathbf{W}_{\text{MWF}}^H\mathbf{V} + L_v|^2\}}. \quad (30)$$

One can identify two extreme cases for the $SNR_{\text{MWF}}^{\text{leakage}}$ as

$$\lim_{G\to\infty} SNR_{\text{MWF}}^{\text{leakage}} = \rho = SNR_{\text{MWF}}^{\text{no-leakage}} \quad (31)$$

$$\lim_{G\to 0} SNR_{\text{MWF}}^{\text{leakage}} = \frac{P_{L_x}}{P_{L_v}} = SNR_{\text{leakage}} \quad (32)$$

Obviously, for large gains the output SNR is equal to the output SNR of the MWF without leakage. Whereas for small gains the output SNR is equal to the SNR of the signal leakage, i.e., the MWF has no effect on the signal reaching the error microphone [7].

5.2. FF ANC algorithm

The output SNR of the FF ANC algorithm is given by

$$SNR_{\text{FF}} = \frac{\mathcal{E}\{|GC\mathbf{W}_{\text{FF}}^H\mathbf{X} + L_x|^2\}}{\mathcal{E}\{|GC\mathbf{W}_{\text{FF}}^H\mathbf{V} + L_v|^2\}}. \quad (33)$$

Since for large gains the filter $\mathbf{W}_{\text{FF}}$ reduces to $\mathbf{W}_{\text{MWF}}$, (cf. (13)), the output SNR is equivalent to the output SNR of the MWF without leakage, i.e.,

$$\lim_{G\to\infty} SNR_{\text{FF}} = \rho = SNR_{\text{MWF}}^{\text{no-leakage}}. \quad (34)$$

In this case the signal leakage has no effect on the signal delivered to the error microphone.

For small gains it is unfortunately not possible to derive a simple formula for the output SNR of the FF ANC algorithm (without any other assumptions).

5.3. FF-FB ANC algorithm

Applying the matrix inversion lemma, the filter in (22) becomes

$$\begin{aligned}\widetilde{\mathbf{W}}_{\text{FF-FB}} &= \widetilde{\mathbf{W}}_{\text{MWF}} - (GC^*)^{-1}\mathbf{e}_{\text{M+1}} \\ &= P_s\frac{\widetilde{\mathbf{R}}_v^{-1}\widetilde{\mathbf{H}}}{1+\widetilde{\rho}}H_1^* - (GC^*)^{-1}\mathbf{e}_{\text{M+1}}\end{aligned} \quad (35)$$

with

$$\widetilde{\rho} = P_s\widetilde{\mathbf{H}}^H\widetilde{\mathbf{R}}_v^{-1}\widetilde{\mathbf{H}} \qquad \text{and} \qquad \widetilde{\mathbf{H}} = \begin{bmatrix}\mathbf{H} \\ H_{\text{M+1}}\end{bmatrix}. \quad (36)$$

The output SNR is then given by

$$\begin{aligned}SNR_{\text{FF-FB}} &= \frac{\mathcal{E}\{|GC\widetilde{\mathbf{W}}_{\text{FF-FB}}^H\widetilde{\mathbf{X}} + L_x|^2\}}{\mathcal{E}\{|GC\widetilde{\mathbf{W}}_{\text{FF-FB}}^H\widetilde{\mathbf{V}} + L_v|^2\}} \\ &= \frac{\mathcal{E}\{|GC\widetilde{\mathbf{W}}_{\text{MWF}}^H\widetilde{\mathbf{X}}|^2\}}{\mathcal{E}\{|GC\widetilde{\mathbf{W}}_{\text{MWF}}^H\widetilde{\mathbf{V}}|^2\}} = \widetilde{SNR}_{\text{MWF}}^{\text{no-leakage}} = \widetilde{\rho}.\end{aligned} \quad (37)$$

The FF-FB ANC algorithm delivers a constant output SNR for any amplification gain G. The output SNR is then equal to the output SNR of the MWF, which uses both (external and internal) microphones as input signals.

Since when more microphones are used, the higher the output SNR becomes [8], the FF-FB ANC algorithm yields the highest output SNR, i.e., $\widetilde{\rho} > \rho$ (cf. experimental results in Section 6.2).

6. EXPERIMENTAL RESULTS

6.1. 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 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 [9] ($f_s = 16$ kHz).

The first $L_c = 128$ taps of the measured secondary path C have been considered. The signals were processed using an overlap-add method with a block size of 256 samples and an overlap of 75% between blocks. The correlation matrices $\mathbf{R}_y$, $\mathbf{R}_v$ and $\mathbf{R}_x$ are estimated as

$$\mathbf{R}_y = \frac{1}{N_y} \sum_{i=1}^{N_y} \mathbf{Y}(k,i)\mathbf{Y}^H(k,i) \quad \text{speech present} \tag{38}$$

$$\mathbf{R}_x = \frac{1}{N_y} \sum_{i=1}^{N_y} \mathbf{X}(k,i)\mathbf{X}^H(k,i) \quad \text{speech present} \tag{39}$$

$$\mathbf{R}_v = \frac{1}{N_v} \sum_{i=1}^{N_v} \mathbf{V}(k,i)\mathbf{V}^H(k,i) \quad \text{speech absent} \tag{40}$$

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).

In order to quantify the broadband performance, the speech intelligibility-weighted output SNR [10] has been used, which takes into account the band importance function I_k, i.e.,

$$\text{SNR}_{int} = \sum_{k=1}^{K} I_k \text{SNR}_k, \tag{41}$$

where SNR_k is the output SNR in the kth frequency band (which has been theoretically analysed in section 5).

6.2. Results

Figure 3 depicts the output SNR_{int} at the error microphone for the MWF (with/without leakage) and for the ANC-motivated algorithms, where the broadband gain G varies from 0 dB to 50 dB.

For large hearing aid amplification gains the output SNR of the MWF is equal to the output SNR of the MWF without leakage (cf. (31)), whereas for small gains the performance of the MWF is degraded by the signal leakage.

Since for large gain values the filter of the FF ANC algorithm reduces to the MWF solution (cf. (13)), the output SNR is equal to the output SNR of the MWF without leakage (cf. (34)), i.e., the signal leakage has no effect on the signal delivered to the error microphone.

The FF-FB ANC algorithm delivers an almost constant output SNR. This supports the theoretical results i.e., the FF-FB ANC algorithm compensates the signal leakage (cf. (24)) and has a constant output SNR independent of the amplification gain G (cf. (37)).

Furthermore, the FF-FB ANC yields the highest output SNR. This can be explained by the fact that in this case the signal leakage is used as an additional input signal, i.e., information from more microphone signals is being used.

7. CONCLUSION

For a single speech source we have theoretically shown that for large gains the standard MWF and the FF ANC algorithms deliver the same output SNR, which is equal to the output SNR of the MWF without leakage.

Moreover we have shown that the error microphone signal of the FF-FB ANC is independent of the signal leakage L_y and can be interpreted as the output signal of the MWF, which uses both external and internal microphones as input signals. The proposed

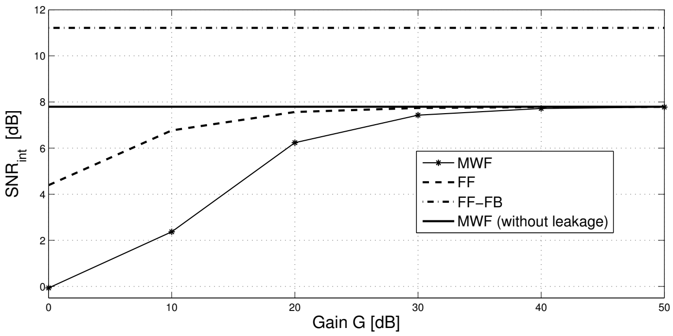

Fig. 3. Speech intelligibility-weighted output SNR of the MWF (with/without leakage) and ANC-motivated algorithms. The speech intelligibility-weighted input SNR in the first microphone was 0 dB.

FF-FB ANC algorithm delivers the highest output SNR, which is then independent of the hearing aid amplification gain G.

In future work we will analyse the effect of the estimation errors in the secondary path C on the performance of the algorithms.

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