A WEIGHTED APPROACH FOR INTEGRATED ACTIVE NOISE CONTROL AND NOISE REDUCTION IN HEARING AIDS

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ABSTRACT

This paper presents a weighted approach for integrated active noise control and noise reduction in hearing aids. An integrated scheme has been introduced previously to tackle secondary path effects and effects of noise leakage through an open fitting. This scheme, however, does not allow to balance between the noise reduction and the active noise control. In some circumstances it will indeed be useful to emphasize one of the functional blocks. A scheme based on a weighted mean squared error criterion is presented here and compared experimentally with the original un-weighted scheme.

1. INTRODUCTION

The usage of hearing aids with an open fitting has become more common over the past years mainly owing to the availability of more efficient feedback control schemes and fast signal processing units. Whereas removing the earmold reduces the occlusion effect and improves the physical comfort [10], one major drawback is that the noise leakage through the fitting cannot be neglected anymore.

One efficient way to cancel this undesired noise leakage is to use Active Noise Control (ANC) [6][11]. In the hearing aids framework, ANC then has to be performed together with a Noise Reduction (NR) algorithm [5][9]. A scheme integrating the two functional blocks and based on a filtered-x [2][3][15] version of the Multichannel Wiener Filter (MWF) algorithm (the so-called FxMWF) has been introduced in [13][14]. The objectives of this algorithm are to attenuate the noise component of the leakage (i.e. ANC) and to minimize the difference between an unknown desired speech signal and the signal delivered at the tympanic membrane (i.e. NR), the balance between these two objectives being fixed. In some cases however, it will be useful to adjust the trade-off between ANC and NR, e.g. when the input signal does not contain any speech or when the ANC is found to be inefficient.

The concept of weighted NR has been introduced in [7] and later applied in the MWF framework to derive the so-called Speech Distortion Weighted Multichannel Wiener Filter (SDW-MWF) [4][5][12]. A similar approach is used here to derive a weighted version of the integrated ANC and NR scheme based on FxMWF. The weighted scheme then allows to emphasize either the ANC or the NR providing an improved signal-to-noise ratio (SNR) or a lower speech distortion (SD) depending of the weight applied.

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This paper will present a perfomance comparison between the original un-weighted integrated ANC and NR scheme and the weighted version formulated here, both of them based on FxMWF and applied in hearing aids with an open fitting. The signal model and the un-weighted integrated ANC and NR are described in Section 2. Section 3 introduces the weighted approach for integrated ANC and NR. Experimental results are presented in Section 4 and finally Section 5 presents a summary of the paper.

2. PROBLEM STATEMENT

Speech enhancement in hearing aids is based on standard NR techniques ignoring the effects of noise leakage through the fitting and the secondary path between the loudspeaker and the tympanic membrane. The leakage signal is not processed in the hearing aid therefore it is not possible to improve its SNR using standard NR algorithms. It is possible however to attenuate the leakage signal's noise component using ANC. This section introduces the signal model and notation, and reviews the integrated ANC and NR scheme presented in [13][14].

2.1 Signal model

Let M be the number of microphones (channels). The signal x_m for microphone m has a desired speech part x_m^s and an additive noise part x_m^n, i.e.:

$$x_m[k] = x_m^s[k] + x_m^n[k] \quad m \in \{1 \dots M\} \tag{1}$$

where k is the time index.

In the sequel, superscripts s and n will also be used for other signals and vectors, to denote their speech and noise component, respectively. Signal model (1) holds for so-called "speech plus noise periods". There are also "noise only periods" (i.e. speech pauses), during which only a noise component is observed.

The column vector $\mathbf{x}_m[k]$ contains the N last samples of channel m:

$$\mathbf{x}_m[k] = [x_m[k] \dots x_m[k-N+1]]^T \tag{2}$$

The compound vector gathering all channels is:

$$\mathbf{x}^T[k] = [\mathbf{x}_1^T[k] \dots \mathbf{x}_M^T[k]] \tag{3}$$

An optimal (Wiener) filter $\mathbf{w}^T[k] = [\mathbf{w}_1^T[k] \dots \mathbf{w}_M^T[k]]$ (of length N) will be designed and applied to the signals, which minimizes a Mean Squared Error (MSE) criterion:

$$J_{MSE}[k] = E\{|e[k]|^2\} \tag{4}$$

Here $e[k]$ is an error signal to be defined next, depending on the scheme applied.
The filter output signal $z[k]$ is defined as:

$$z[k] = \mathbf{w}^T[k]\mathbf{x}[k] \tag{5}$$

which will be the hearing aid output signal, fed to the loudspeaker.

2.2 Integrated active noise control and noise reduction

The algorithm introduced in [13][14] integrates both the NR and an ANC in a single set of adaptive filters (fig. 1). It combines NR with secondary path and leakage compensation:

The so-called secondary path represents the propagation from the loudspeaker to the tympanic membrane (including the loudspeaker response itself). As in any ANC scheme, it has to be taken into account explicitly. Assuming that the loudspeaker characteristic is approximately linear, the secondary path can be represented by the transfer function $C(z)$.

The purpose of the NR is to provide an optimal estimate of a desired signal $d_{NR}[k]$, which is chosen to be equal to the (unknown) speech component in the first microphone, up to a delay and amplified by a gain G, representing hearing loss compensation:

$$d_{NR}[k] = G \cdot x_1^s[k-\Delta] \tag{6}$$

The aim is to deliver this desired signal at the tympanic membrane in spite of the secondary path.

Finally, the purpose of the ANC is to cancel the noise component $l^n[k]$ of the leakage signal $l[k]$ arriving at the tympanic membrane through the open fitting. In the hearing aids context, the speech component of the leakage signal can provide cues which, e.g., are helpful for speaker localization. Therefore, it is chosen here to cancel only the noise component of the leakage signal and preserve the speech component.

The overall desired signal (at the tympanic membrane) to be used is then:

$$\begin{aligned} d_{Int}[k] &= -l^n[k] + G \cdot x_1^s[k-\Delta] \\ &= -l^n[k] + d_{NR}[k] \end{aligned} \tag{7}$$

Hence the MSE criterion to be minimized is:

$$J_{MSE}[k] = E\{|e_{Int}[k]|^2\} \tag{8}$$
$$e_{Int}[k] = C(z) * \mathbf{w}^T[k]\mathbf{x}[k] + l^n[k] - d_{NR}[k] \tag{9}$$

The algorithm relies on a filtered-x type operation based on an estimate $\hat{C}(z)$ of the secondary path $C(z)$.

$$\hat{\mathbf{c}} = [\hat{c}_0 \ldots \hat{c}_{L-1}]^T \tag{10}$$

The filtered reference signals are:

$$\begin{aligned} y_m[k] &= \hat{\mathbf{c}}^T[x_m[k] \ldots x_m[k-L+1]]^T \quad m \in \{1 \ldots M\} \\ \mathbf{y}_m[k] &= [y_m[k] \ldots y_m[k-N+1]]^T \\ \mathbf{y}^T[k] &= [\mathbf{y}_1^T[k] \ldots \mathbf{y}_M^T[k]] \end{aligned} \tag{11}$$

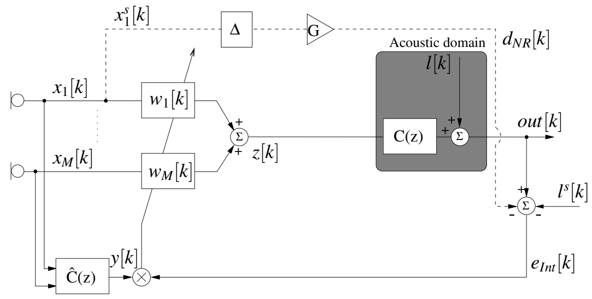

Figure 1: Integrated Active Noise Control and Noise Reduction system

Assuming that the secondary path identification error is small ($\hat{C}(z) \approx C(z)$) and that the filter $\mathbf{w}$ is adapting slowly, the MSE criterion (8) can be written as follows:

$$J_{MSE}[k] \approx E\{|\mathbf{w}^T[k]\mathbf{y}[k] + l^n[k] - d_{NR}[k]|^2\} \tag{12}$$

The optimal filter (FxMWF) minimizing (8) is then:

$$\mathbf{w}[k] = \mathbf{R}_{yy}^{-1}[k]\mathbf{r}_{yd_{Int}}[k] \tag{13}$$

Here $\mathbf{R}_{yy}[k]$ is the correlation matrix of the filtered reference signal $\mathbf{y}[k]$ and $\mathbf{r}_{yd_{Int}}[k]$ is the cross-correlation vector between the filtered reference signal $\mathbf{y}[k]$ and the desired signal $d_{Int}[k]$.

Note that by assuming that the speech and noise components of the input signals are uncorrelated the cross-correlation vector can be estimated using:

$$\begin{aligned} \mathbf{r}_{yd_{Int}}[k] &= \mathbf{r}_{y^s d_{NR}}[k] - \mathbf{r}_{y^n l^n}[k] \\ &= \mathbf{r}_{yx_{1,\Delta}}[k] - G \cdot \mathbf{r}_{y^n x^n_{1,\Delta}}[k] - \mathbf{r}_{y^n l^n}[k] \end{aligned} \tag{14}$$

with

$$\mathbf{r}_{yx_{1,\Delta}}[k] = E\{\mathbf{y}[k]x_1[k-\Delta]\} \tag{15}$$
$$\mathbf{r}_{y^n x^n_{1,\Delta}}[k] = E\{\mathbf{y}^n[k]x_1^n[k-\Delta]\} \tag{16}$$
$$\mathbf{r}_{y^n l^n}[k] = E\{\mathbf{y}^n[k]l^n[k]\} \tag{17}$$

While $\mathbf{R}_{yy}[k]$ and $\mathbf{r}_{yx_{1,\Delta}}[k]$ are estimated during speech plus noise periods, $\mathbf{r}_{y^n x_1^n[k-\Delta]}$ and $\mathbf{r}_{y^n l^n}[k]$ can be estimate during noise only periods with:

$$l^n[k] \approx e^n_{Int}[k] - \mathbf{w}^T[k]\mathbf{y}^n[k] \tag{18}$$

The above algorithm has been derived and evaluated in [13][14].

Assuming that the noise and speech components are uncorrelated in (12), the MSE criterion can be written as follows:

$$\begin{aligned} J_{MSE}[k] \approx\ & E\{|\mathbf{w}^T[k]\mathbf{y}^s[k] - d_{NR}[k]|^2\} \\ & + E\{|\mathbf{w}^T[k]\mathbf{y}^n[k] + l^n[k]|^2\} \end{aligned} \tag{19}$$

The first term of the right hand side of (19) corresponds to the difference between the unkown desired speech signal and the speech component of the signal delivered at the tympanic membrane, i.e. the speech distortion (SD). The second term specifies the noise sound pressure at the tympanic membrane and thus corresponds to the ANC.

The un-weighted integrated scheme thus minimizes an MSE criterion (19) which can be viewed as the sum of an ANC term and an SD term. Therefore, it may exhibit lower noise attenuation performance than an ANC filter alone, minimizing the MSE criterion (20). On the other hand, when the leakage signal is dominant (e.g. for low gains G), the ANC term dominates the SD term in the optimization, eventually leading to a large SD. Therefore, the un-weighted integrated scheme is found to introduce more SD than a standard NR scheme minimizing the MSE criterion (21).

$$\begin{aligned} E\{|e_{ANC}|^2\} &= E\{|C * \mathbf{w}^T[k]\mathbf{x}^n[k] + l^n[k]|^2\} \quad (20)\\ E\{|e_{NR}|^2\} &= E\{|C * \mathbf{w}^T[k]\mathbf{x}[k] - d_{NR}[k]|^2\} \quad (21)\end{aligned}$$

When the input signal does not contain any speech, the NR is not needed and so ANC alone can perform better than the un-weighted integrated scheme. Also, if the background noise is, e.g., high-frequency noise when typically the ANC is found to be inefficient, using a NR alone may reduce the SD introduced by the un-weighted integrated algorithm (see also Section 4).

The MSE criterion (8) which is minimized by the un-weighted integrated scheme, cannot be decomposed in the sum of an ANC term and an NR term. It is therefore not possible to derive straightforwardly a scheme that balances between the ANC and the NR from the un-weigthed scheme.

3. WEIGHTED ACTIVE NOISE CONTROL AND NOISE REDUCTION

The algorithm described in this section applies a different weight to the ANC objective (20) and to the NR objective (21). Therefore, unlike for the un-weighted scheme described in previous section, the two MSE criteria have to be computed explicitly. A weight is then applied on one of the criteria, allowing to emphasize the effect of one of the functional blocks.

The problem can be seen as trying to minimize the residual noise at the tympanic membrane (i.e. ANC) under the constraint that the difference between the desired signal and the filtered signal, as delivered to the tympanic membrane (i.e. NR), is kept below a given threshold:

$$\min_{\mathbf{w}} E\{|e_{ANC}|^2\}, \text{ subject to } E\{|e_{NR}|^2\} \leq T \quad (22)$$

Introducing the Lagrange-multiplier $\mu > 0$, the MSE criterion to be minimized is then :

$$J_{MSE}[k] = \mu E\{|e_{NR}|^2\} + E\{|e_{ANC}|^2\} \quad (23)$$

The factor $\mu \in [0, \infty]$ trades-off between the NR and the ANC. When $\mu \to 0$, the MSE in (23) reduces to (20). The system tends to behave like a standard ANC algorithm. The algorithm then achieves high noise attenuation performance but it also introduces extensive SD, as the speech component is not taken into account in the optimization process.

When $\mu \to \infty$, the MSE in (23) reduces to (21). The system then behaves as a MWF-based NR algorithm. The algorithm introduces less SD but the noise attenuation performance is decreased.

In practice, to compute the MSE in (23) the two error signals ($e_{ANC}[k]$ and $e_{NR}[k]$) have to be computed explicitly. Assuming that the secondary path identification error is small ($\hat{C}(z) \approx C(z)$) and that the filter $\mathbf{w}$ is adapting slowly, these error signals can be written as follows:

$$\begin{aligned} e_{ANC}[k] &\approx \mathbf{w}^T[k]\mathbf{y}^n[k] + l^n[k] \quad (24)\\ e_{NR}[k] &\approx \mathbf{w}^T[k]\mathbf{y}[k] - d_{NR}[k] \quad (25)\end{aligned}$$

Assuming that the noise and the speech components in (25) are uncorrelated, the MSE criteria (20) and (21) can be rewritten as follows:

$$\begin{aligned} E\{|e_{ANC}|^2\} &\approx E\{|\mathbf{w}^T[k]\mathbf{y}^n[k] + l^n[k]|^2\} \quad (26)\\ E\{|e_{NR}|^2\} &\approx E\{|\mathbf{w}^T[k]\mathbf{y}^s[k] - d_{NR}[k]|^2\}\\ &\quad + E\{|\mathbf{w}^T[k]\mathbf{y}^n[k]|^2\} \quad (27)\end{aligned}$$

The optimal filter (FxMWF) which minimizes the MSE criterion (23) is then:

$$\mathbf{w}_\mu[k] = \mathbf{R}_\mu^{-1}[k]\mathbf{r}_\mu[k] \quad (28)$$

Here $\mathbf{R}_\mu[k]$ is the weighted correlation matrix of the filtered reference signal $\mathbf{y}[k]$ and $\mathbf{r}_\mu[k]$ is the weighted cross-correlation vector between the filtered reference signal $\mathbf{y}[k]$ and the desired signal $d_{Int}[k]$:

$$\begin{aligned} \mathbf{R}_\mu[k] &= \mu \mathbf{R}_{y^s y^s}[k] + (1+\mu)\mathbf{R}_{y^n y^n}[k] \quad (29)\\ \mathbf{r}_\mu[k] &= \mu \mathbf{r}_{y^s d_{NR}}[k] - \mathbf{r}_{y^n l^n}[k] \quad (30)\end{aligned}$$

Note that by assuming once again that the speech and the noise components of the input signals are uncorrelated the correlation matrix and the cross-correlation vector can be estimated using:

$$\begin{aligned} \mathbf{R}_\mu[k] &= \mu \mathbf{R}_{yy}[k] + \mathbf{R}_{y^n y^n}[k] \quad (31)\\ \mathbf{r}_{y^s d_{NR}}[k] &= G \cdot (\mathbf{r}_{yx_{1,\Delta}}[k] - \mathbf{r}_{y^n x^n_{1,\Delta}}[k]) \quad (32)\end{aligned}$$

with

$$\begin{aligned} \mathbf{R}_{yy}[k] &= E\{\mathbf{y}[k]\mathbf{y}^T[k]\} \quad (33)\\ \mathbf{R}_{y^n y^n}[k] &= E\{\mathbf{y}^n[k]\mathbf{y}^{nT}[k]\} \quad (34)\end{aligned}$$

and $\mathbf{r}_{yx_{1,\Delta}}[k]$, $\mathbf{r}_{y^n x^n_{1,\Delta}}$ and $\mathbf{r}_{y^n l^n}[k]$ are defined in (15), (16) and (17) respectively.

While $\mathbf{R}_{yy}[k]$ and $\mathbf{r}_{yx_{1,\Delta}}[k]$ are estimated during speech plus noise periods, $\mathbf{R}_{y^n y^n}[k]$, $\mathbf{r}_{y^n x^n_{1,\Delta}}$ and $\mathbf{r}_{y^n l^n}[k]$ can be estimated during noise only periods with:

$$l^n[k] \approx e^n_{ANC}[k] - \mathbf{w}^T_\mu[k]\mathbf{y}^n[k] \quad (35)$$

The expression of the optimal filter can then be simplified by substituting (31) and (32) in (28):

$$\mathbf{w}_\mu[k] = (\mu \mathbf{R}_{yy}[k] + \mathbf{R}_{y^n y^n}[k])^{-1}(\mu \mathbf{r}_{y^s d_{NR}}[k] - \mathbf{r}_{y^n l^n}[k]) \quad (36)$$

From (36) it appears cleary that the two extreme cases for the filter $\mathbf{w}_\mu[k]$ are given by:

$$\begin{aligned} \lim_{\mu \to 0} \mathbf{w}_\mu[k] &= -\mathbf{R}_{y^n y^n}[k]^{-1}\mathbf{r}_{y^n l^n}[k] \quad (37)\\ \lim_{\mu \to \infty} \mathbf{w}_\mu[k] &= \mathbf{R}_{yy}[k]^{-1}\mathbf{r}_{y^s d_{NR}}[k] \quad (38)\end{aligned}$$

Here (37) is the expression of an ANC filter which minimizes the noise pressure at the tympanic membrane and (38) is the expression of a NR filter which also compensates for the secondary path. The filter described in (36) therefore integrates the two functional blocks with the coefficient μ used as a trade-off parameter between the NR and the ANC.

4. EXPERIMENTAL RESULTS

The algorithm introduced in Section 3 has been tested experimentally and its performance has been compared with the performance of the un-weighted integrated algorithm [13][14] described in Section 2.2.

4.1 Experimental setup

The simulations were run for a two-microphone behind-the-ear (BTE) hearing aid, with a speech source at 0° and a babble noise source at 270°. The BTE is worn on the left ear, facing the noise source. The SNR for the source signals is set to $5dB$. Three performance measures are considered here: the noise attenuation at the tympanic membrane, the SD and the SNR of the output signal.

The noise attenuation improvement is defined as

$$\Delta POW = POW_{weigth} - POW_{unweigth} \tag{39}$$

where POW_{weigth} and $POW_{unweigth}$ are the broadband power (in dB) of the noise signal at the tympanic membrane obtained with the weighted scheme and with the un-weighted scheme, respectively.

The speech distortion improvement is defined as

$$\Delta SD = SD_{intellig,weigth} - SD_{intellig,unweigth} \tag{40}$$

where $\mathrm{SD}_{intellig,weigth}$ and $\mathrm{SD}_{intellig,unweigth}$ represent the output SD (in dB) for the weighted scheme and for the un-weighted scheme, respectively.

An intelligibility weighted spectral distortion measure is used defined as

$$SD_{intellig} = \sum_i I_i SD_i \tag{41}$$

where I_i is the band importance function defined in [1] and SD_i the average spectral distortion (in dB) in the i-th one third octave band,

$$SD_i = \frac{1}{(2^{1/6} - 2^{-1/6}) f_i^c} \int_{2^{-1/6} f_i^c}^{2^{1/6} f_i^c} |10 \log_{10} G^s(f)| df \tag{42}$$

with center frequencies f_i^c and $G^s(f)$ the squared magnitude of the transfer function for the speech component from the input to the output of the noise reduction algorithm.

The intelligibility-weighted signal-to-noise ratio (SNR) [8] is used here to compute the SNR improvement which is defined as

$$\Delta SNR_{intellig} = \sum_i I_i (SNR_{i,weigth} - SNR_{i,unweigth}) \tag{43}$$

where I_i is the band importance function and $\mathrm{SNR}_{i,weigth}$ and $\mathrm{SNR}_{i,unweigth}$ represent the output SNR (in dB) of the ith band, for the weighted scheme and the un-weighted scheme, respectively.

The influence of the weighting factor μ on the noise attenuation at the tympanic membrane and on the SD of the output signal is first examined, the output SNR improvement is then considered.

4.2 Noise power and speech distortion

To derive the filter introduced in Section 3 the MSE criterion has been explicitly decomposed as the sum of two criteria, and a weight is applied to emphazises one of the functional blocks (ANC or NR). Figs. 2 and 3 present the noise attenuation improvement (39) and the SD improvement (40), for the weighted scheme compared against the un-weighted scheme as a function of μ and for different values of the gain G.

When $\mu \to 0$, the weighted scheme is attenuating the noise at the tympanic membrane more efficiently than the un-weighted scheme (fig.2), i.e. it behaves as an ANC algorithm. This also means that the algorithm introduces a lot of SD (fig.3).

When μ increases, the noise attenuation improvement vanishes (fig.2) while the SD decreases (fig.3). When $\mu \to \infty$, the weighted scheme behaves as a standard NR and some improvement can be done on the SD performance compared against the fixed un-weighted scheme.

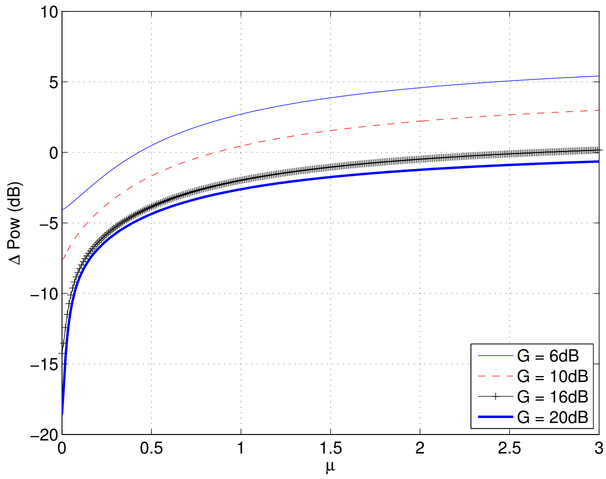

Figure 2: Noise attenuation improvement compared to the integrated scheme

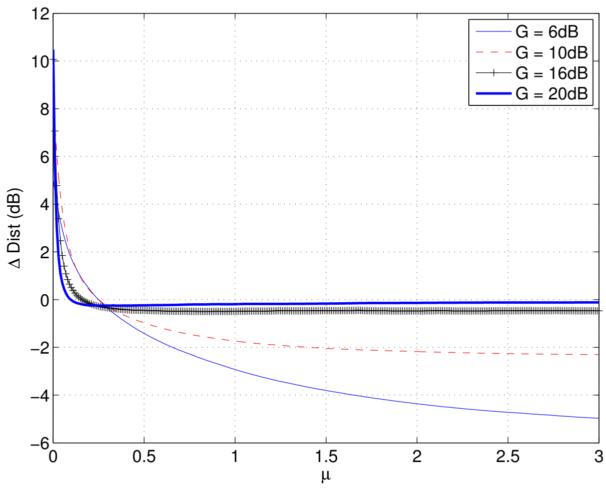

Figure 3: SD improvement compared to the integrated scheme

4.3 SNR performance of the weighted scheme

The position of the sources and the SNR for the source signals leads to a so-called leakage SNR (which corresponds to the SNR when the hearing aid is turned off) equal to $-1.3dB$. For all values of the gain, the integrated scheme provides an SNR improvement of about $10dB$ [13][14]. Fig. 4 shows the SNR improvement (43) of the weighted scheme compared against the SNR performance of the un-weighted scheme as a function of μ and for different values of the gain G.

For low values of the weighting coefficient (up to around 0.5), the weighted scheme provides an SNR improvement which can be $4dB$ higher than the SNR improvement obtained with the un-weighted scheme. When μ increases, the weighted scheme behaves more like a standard NR scheme, then the un-weighted scheme exhibits a better SNR performance.

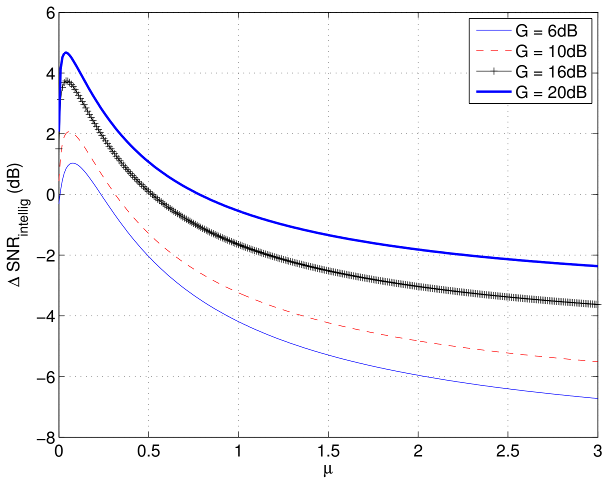

Figure 4: SNR improvement compared to the integrated scheme

5. CONCLUSION

This paper presented a weighted approach for integrated ANC and NR in hearing aids to emphazise one of the functional blocks. When the signal does not contain any speech, the weighted scheme allows to focus on ANC and minimize the power of the noise signal at the tympanic membrane. On the other hand, if the ANC is found to be inefficient for the considered background noise scenario the emphasis can be put on the NR, and then the weighted scheme exhibits improved SD performance compared to an un-weighted integrated scheme.

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