

Output SNR analysis of integrated active noise control and noise reduction in hearing aids under a single speech source scenario¹

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Abstract

This paper analyses the output signal-to-noise ratio for a standard noise reduction scheme and for an integrated active noise control and noise reduction scheme, both applied in the hearing aid framework including the effects of signal leakage through an open fitting and secondary path effects.

In particular, a standard noise reduction scheme based on the multichannel Wiener filter and an integrated active noise control and noise reduction scheme based on the filtered-X multichannel Wiener filter are analysed and their signal-to-noise ratio are derived under a single speech source scenario.

OUTPUT SNR ANALYSIS OF INTEGRATED ACTIVE NOISE CONTROL AND NOISE REDUCTION IN HEARING AIDS UNDER A SINGLE SPEECH SOURCE SCENARIO

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ABSTRACT

This paper analyses the output signal-to-noise ratio for a standard noise reduction scheme and for an integrated active noise control and noise reduction scheme, both applied in the hearing aid framework including the effects of signal leakage through an open fitting and secondary path effects.

In particular, a standard noise reduction scheme based on the multichannel Wiener filter and an integrated active noise control and noise reduction scheme based on the filtered-X multichannel Wiener filter are analysed and their signal-to-noise ratio are derived under a single speech source scenario.

Index Terms— Active noise control, multichannel Wiener filter, noise reduction, hearing aids, signal-to-noise ratio.

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 [7], one major drawback is that the signal leakage through the fitting cannot be neglected anymore. Conventional Noise Reduction (NR) systems such as the Generalized Sidelobe Canceller (GSC) [6] or techniques based on the Multichannel Wiener Filter (MWF) [3] do not take this contribution into account. Combined with the attenuation in the acoustic path between the sound source (hearing aid loudspeaker) and the tympanic membrane (the so-called secondary path), the noise leaking through the fitting can override the action of the processing done in the hearing aid.

It has been shown that integrating Active Noise Control (ANC) [4, 8, 5] and NR, based on the so-called Filtered-X Multichannel Wiener Filter (FxMWF) [10], is an efficient

way to cancel undesired leakage signal and restore the standard NR performance [11].

In a single speech source scenario it is possible to compute the output signal-to-noise ratio (SNR) of MWF-based NR schemes [1]. In this paper, the single speech source assumption is applied to derive the output SNR of an MWF-based NR scheme in the context of hearing aids with an open fitting, when signal leakage and the secondary path are taken into account. The single speech source assumption is then also used to derive the output SNR performance of a frequency-domain implementation of the integrated ANC and NR scheme, when the number of sound sources (speech plus noise sources) does not exceed the number of input microphones. This allows to verify the results from the simulations done in previous works [11].

The signal model for the single speech source scenario is described in Section 2. The SNR of MWF-based NR and the effects of the signal leakage and the secondary path are commented on in Section 3. Section 4 analyses the SNR of the integrated ANC and NR scheme. Finally Section 5 presents the conclusions of this paper.

2. SIGNAL MODEL

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

$$X_m(\omega) = X_m^s(\omega) + X_m^n(\omega) \quad m \in \{1 \dots M\} \quad (1)$$

where $\omega = 2\pi f$ is the frequency-domain variable. For conciseness, ω will be omitted in all subsequent equations.

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 compound vector gathering all channels is:

$$\mathbf{X}^T = [X_1 \dots X_M] \quad (2)$$

An optimal (Wiener) filter $\mathbf{W}^T = [W_1 \dots W_M]$ will be designed and applied to the signals, which minimizes a Mean Squared Error (MSE) criterion:

$$J_{MSE} = \mathbb{E}\{|E|^2\} \quad (3)$$

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Here E is an error signal to be defined next, depending on the scheme applied.

The filter output signal Z (*i.e.*, the signal to be fed in the hearing aid loudspeaker) is defined as:

$$Z = \mathbf{W}^H \mathbf{X} \quad (4)$$

where H denotes the Hermitian transpose.

The desired speech signal, as defined in [11], is arbitrarily chosen to be the (unknown) speech component of the first microphone signal ($m = 1$), up to a delay Δ . This can be written:

$$D_{NR} = \mathbf{G}_{1,\Delta}^H \mathbf{X}^s \quad (5)$$

$$\mathbf{G}_{1,\Delta} = [Ge^{-j\omega\Delta} \ 0 \dots 0] \quad (6)$$

$$= G\mathbf{e}_{1,\Delta} \quad (7)$$

where the gain G is the amplification that compensates for the hearing loss.

The autocorrelation matrices of the speech component and the noise component of the microphones signals are respectively given by:

$$\mathbf{R}_s = \mathbb{E}\{\mathbf{X}^s \mathbf{X}^{sH}\} \quad (8)$$

$$\mathbf{R}_n = \mathbb{E}\{\mathbf{X}^n \mathbf{X}^{nH}\} \quad (9)$$

If the speech signal and the noise signal are assumed to be uncorrelated, \mathbf{R}_n can be estimated during "noise only periods" and \mathbf{R}_s can be estimated during "speech plus noise periods" using:

$$\mathbf{R}_s = \mathbb{E}\{\mathbf{X}^H \mathbf{X}\} - \mathbf{R}_n \quad (10)$$

It is assumed in all subsequent schemes that only one speech source is present. The autocorrelation matrix of the speech component of the microphone signal (\mathbf{R}_s) is then rank-1 and can be rewritten as:

$$\mathbf{R}_s = P_s \mathbf{A} \mathbf{A}^H \quad (11)$$

where P_s is the power of the speech signal and \mathbf{A} is the M -dimensional steering vector which contains the acoustic transfer functions from the speech source to the behind-the-ear hearing aid microphones.

3. MWF-BASED NOISE REDUCTION

The noise reduction scheme based on MWF is designed to minimize the squared distance between the filtered microphone signal ($\mathbf{W}^H \mathbf{X}$) and the desired speech signal ($D_{NR} = \mathbf{X}^s \mathbf{G}_{1,\Delta}$). Therefore, the Mean Square Error criterion to be minimized is:

$$J_{MSE} = \mathbb{E}\{|E_{NR}|^2\} \quad (12)$$

$$E_{NR} = \mathbf{W}^H \mathbf{X} - \mathbf{G}_{1,\Delta}^H \mathbf{X}^s \quad (13)$$

The corresponding Wiener filter is:

$$\mathbf{W}_{NR} = (\mathbf{R}_s + \mathbf{R}_n)^{-1} \mathbf{R}_s \mathbf{G}_{1,\Delta} \quad (14)$$

In the single speech source case, the Woodbury identity can be applied to compute the inverse of the pencil matrix ($\mathbf{R}_s + \mathbf{R}_n$) [1]. The filter (14) can then be rewritten as follows:

$$\mathbf{W}_{NR} = \frac{\mathbf{R}_n^{-1}}{\rho + 1} \mathbf{R}_s \mathbf{G}_{1,\Delta} \quad (15)$$

with

$$\rho = P_s \mathbf{A}^H \mathbf{R}_n^{-1} \mathbf{A} \quad (16)$$

The filter (15) is designed without taking the effects of the signal leakage and the secondary path effects into account. In this section, the output SNR performance of such a system is first computed when the signal leakage and the secondary path effects are ignored. The effects of these perturbations on the output SNR of the MWF-based NR scheme are then analysed.

3.1. Output SNR

When the leakage signal and the secondary path effects are ignored the output SNR of the system is given by:

$$\begin{aligned} SNR_{NR-noLeakage} &= \frac{\mathbf{W}_{NR}^H \mathbf{R}_s \mathbf{W}_{NR}}{\mathbf{W}_{NR}^H \mathbf{R}_n \mathbf{W}_{NR}} \\ &= \frac{\rho^2}{\rho+1} P_{DNR} \\ &= \frac{\rho}{\rho+1} P_{DNR} \end{aligned} \quad (17)$$

$$= \rho \quad (18)$$

Where P_{DNR} is the power of the desired speech signal:

$$P_{DNR} = \mathbf{G}_{1,\Delta}^H \mathbf{R}_s \mathbf{G}_{1,\Delta} \quad (19)$$

$$= G^2 \sigma_{s_1} \quad (20)$$

and σ_{s_1} is the variance of the speech component in the first microphone signal ($m = 1$):

$$\sigma_{s_1} = \mathbb{E}\{X_1^s X_1^{sH}\} \quad (21)$$

This result is similar to the SNR formula in [1] and does not depend on the amplification gain G . However, it does not include the effects of the secondary path and the signal leakage effects.

3.2. Output SNR under signal leakage effects

In the context of a hearing aid with an open fitting there is no earmold to prevent the ambient sound from leaking into the ear canal, which results in an additional leakage signal L reaching the tympanic membrane [2]. No direct processing can be done on this signal, therefore its SNR is generally lower than for the signal provided by the hearing aid (4).

Taking the signal leakage effects into account, the output SNR of the standard NR scheme can be rewritten:

$$SNR_{NR-Leakage} = \frac{\mathbb{E}\{|\mathbf{W}^H \mathbf{X}^s + L^s|^2\}}{\mathbb{E}\{|\mathbf{W}^H \mathbf{X}^n + L^n|^2\}} \quad (22)$$

$$= \frac{\frac{\rho^2}{\rho+1} P_{DNR} + \frac{\rho}{\rho+1} \alpha + P_{L_s}}{\frac{\rho}{\rho+1} P_{DNR} + \frac{1}{\rho+1} \alpha + P_{L_n}} \quad (23)$$

with

$$P_{L_s} = \mathbb{E}\{L^s L^{sH}\} \quad (24)$$

$$P_{L_n} = \mathbb{E}\{L^n L^{nH}\} \quad (25)$$

$$\alpha = \mathbf{G}_{1,\Delta}^H \mathbf{r}_{sl} + \mathbf{r}_{ls} \mathbf{G}_{1,\Delta} \quad (26)$$

$$= G(\mathbf{e}_{1,\Delta}^H \mathbf{r}_{sl} + \mathbf{r}_{ls} \mathbf{e}_{1,\Delta}) \quad (27)$$

where P_{L_s} and P_{L_n} are the power of the speech component and the noise component of the leakage signal, respectively.

The cross-correlation vectors between the speech component of the microphone signal and the speech component of the leakage signal (\mathbf{r}_{sl} and \mathbf{r}_{ls}) are given by:

$$\mathbf{r}_{sl} = \mathbb{E}\{\mathbf{X}^{sH} L^s\} \quad (28)$$

$$\mathbf{r}_{ls} = \mathbb{E}\{L^{sH} \mathbf{X}^s\} \quad (29)$$

From equations (20), (27) and (23) one can identify the two extreme cases for the output SNR of the filter (15), when the signal leakage is taken into account, which are given by:

$$\lim_{G \rightarrow 0} \mathbf{W}_{NR-Leakage} = \frac{P_{L_s}}{P_{L_n}} = SNR_{leakage} \quad (30)$$

$$\lim_{G \rightarrow \infty} \mathbf{W}_{NR-Leakage} = \rho = SNR_{NR-noLeakage} \quad (31)$$

When the amplification gain G is low, the output SNR is equivalent to the leakage SNR, *i.e.*, the NR has no effect on the signal delivered at the tympanic membrane. When the amplification gain G is high, the output SNR is equivalent to the output SNR of the NR scheme without leakage, *i.e.*, the signal leakage has no effect on the signal reaching the tympanic membrane (fig. 2).

3.3. Output SNR under signal leakage and secondary path effects

The MWF-based NR scheme ignores the so-called secondary path, *i.e.*, the propagation from the loudspeaker to the tympanic membrane (including the loudspeaker response itself). This secondary path however has an effect on the performance of the NR scheme. In the context of hearing aids, this secondary path usually acts as an attenuation. Therefore, the degradations caused by the signal leakage may be even more important when the secondary path is taken into account.

Assuming that the loudspeaker characteristic is approximately linear, the secondary path can be represented by the transfer function C . The output SNR of the MWF-based NR scheme can then be rewritten to take the signal leakage effects and the effects of the secondary path into account:

$$\begin{aligned} SNR_{NR-Leak+Sec} &= \frac{\mathbb{E}\{|C^* \mathbf{W}^H \mathbf{X}^s + L^s|^2\}}{\mathbb{E}\{|C^* \mathbf{W}^H \mathbf{X}^n + L^n|^2\}} \quad (32) \\ &= \frac{|C|^2 \frac{\rho^2}{\rho+1} P_{D_{NR}} + (C + C^*) \frac{\rho}{\rho+1} \alpha + P_{L_s}}{|C|^2 \frac{\rho}{\rho+1} P_{D_{NR}} + (C + C^*) \frac{1}{\rho+1} \alpha + P_{L_n}} \end{aligned}$$

This expression is very similar to (23) except that due to the attenuation caused by the secondary path C , the system needs a higher amplification G to restore the performance back to the performance of the NR scheme when the signal leakage and the secondary path are not present (fig. 2).

In the single speech source scenario, the theory confirms the observations made in [11]. The secondary path and the signal leakage degrade the NR performance when the amplification G is low. One solution to compensate for these perturbations is to integrate ANC with the NR scheme.

4. INTEGRATED ACTIVE NOISE CONTROL AND NOISE REDUCTION

This section analyses the output SNR performance of a frequency-domain version of the integrated scheme introduced in [11]. This scheme integrates NR and ANC in a single set of adaptive filters. Note that in practice, to ensure the causality of the frequency-domain version of the scheme, the filter coefficients are computed in the frequency-domain while the filtering operation is performed in the time-domain (fig 1), in a similar way as presented in [9].

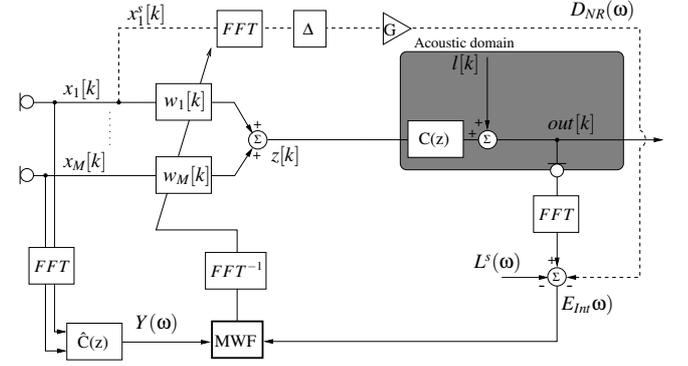


Fig. 1. Integrated ANC and NR

The aim of the integrated scheme is to improve the speech-to-noise ratio, *i.e.*, to minimize the squared distance between a desired speech signal ($D_{NR} = \mathbf{X}^s \mathbf{G}_{1,\Delta}$) and the speech signal delivered to the tympanic membrane ($C^* \mathbf{W}^H \mathbf{X}^s$) and in the same time to minimize the residual noise at the tympanic membrane ($C^* \mathbf{W}^H \mathbf{X}^n + L^n$). Therefore the desired signal to be used here is:

$$D_{Int} = \mathbf{X}^s \mathbf{G}_{1,\Delta} - L^n \quad (33)$$

and the MSE criterion to be minimized is:

$$J_{MSE} = \mathbb{E}\{|E_{Int}|^2\} \quad (34)$$

$$\begin{aligned} E_{Int} &= C \cdot Z - D_{Int} \quad (35) \\ &= C \cdot \mathbf{W}^H \mathbf{X} + L^n - \mathbf{X}^s \mathbf{G}_{1,\Delta} \end{aligned}$$

The optimal filter (FxMWF) minimizing (34) is:

$$\mathbf{W}_{Int} = \frac{\hat{C}^*}{|\hat{C}|^2} (\mathbf{R}_s + \mathbf{R}_n)^{-1} (\mathbf{R}_s \mathbf{G}_{1,\Delta} - \mathbf{r}_{nl}) \quad (36)$$

Where \hat{C} is an estimate of the secondary path C and \mathbf{r}_{nl} is the cross-correlation vector between the noise component of the microphone signal and the noise component of the leakage signal defined as:

$$\mathbf{r}_{nl} = \mathbb{E}\{\mathbf{X}^n L^n\} \quad (37)$$

4.1. Output SNR when the number of sources is less or equal to the number of microphones

When the number of sources (speech plus noise sources) is less or equal to the number of microphones, it can be shown

that the leakage signal can be rewritten as a linear combination of the microphone signals:

$$L = \mathbf{P}^H \mathbf{X} \quad (38)$$

$$\mathbf{P}^T = [P_1 \dots P_M] \quad (39)$$

The filter (36) then becomes:

$$\mathbf{W}_{Int} = \frac{\hat{C}^*}{|\hat{C}|^2} (\mathbf{R}_s + \mathbf{R}_n)^{-1} (\mathbf{R}_s \mathbf{G}_{1,\Delta} - \mathbf{R}_n \mathbf{P}) \quad (40)$$

In the single speech source scenario, the Woodbury identity can be used to invert the pencil matrix, leading to:

$$\mathbf{W}_{Int} = \frac{\hat{C}^*}{|\hat{C}|^2} \left[\frac{\mathbf{R}_n^{-1} \mathbf{R}_s}{1 + \rho} (\mathbf{G}_{1,\Delta} + \mathbf{P}) - \mathbf{P} \right] \quad (41)$$

Assuming that the secondary path identification error is small ($\hat{C} \approx C$), the output SNR of the system is given by:

$$SNR_{Int} = \frac{\mathbb{E}\{|C^* \mathbf{W}^H \mathbf{X}^s + L^s|^2\}}{\mathbb{E}\{|C^* \mathbf{W}^H \mathbf{X}^n + L^n|^2\}} \quad (42)$$

$$= \frac{\mathbb{E}\{|C^* \mathbf{W}^H \mathbf{X}^s + \mathbf{P}^H \mathbf{X}^s|^2\}}{\mathbb{E}\{|C^* \mathbf{W}^H \mathbf{X}^n + \mathbf{P}^H \mathbf{X}^n|^2\}} \quad (43)$$

$$= \frac{\frac{\rho^2}{\rho+1} P_{DNR} + \frac{\rho^2}{\rho+1} \alpha + \frac{\rho^2}{\rho+1} P_{L_s}}{\frac{\rho}{\rho+1} P_{DNR} + \frac{\rho}{\rho+1} \alpha + \frac{\rho}{\rho+1} P_{L_n}} \quad (44)$$

$$= \rho = SNR_{NR-noLeakage} \quad (45)$$

The secondary path and the noise component of the leakage signal are included in the cost function of the integrated scheme (34). Therefore, the signal leakage has no effect on the performance of the system and the scheme delivers a constant output SNR for any amplification gain G . The SNR is then equal to the SNR of the MWF-based NR scheme when the effect of the signal leakage and the secondary path effects are ignored (fig. 2).

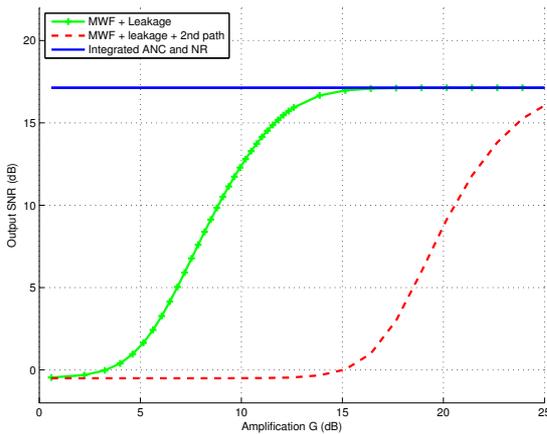


Fig. 2. Output SNR for MWF-based NR scheme and for the integrated ANC and NR scheme

Figure 2 presents the output SNR of the MWF-based NR under signal leakage effects, under signal leakage and secondary path effects and the output SNR of the integrated ANC and NR, all are based on the formulas derived in this paper. The amplification G varies from 0 dB to 25 dB.

5. CONCLUSION

For a single speech source scenario, it can be shown that the signal leakage and the secondary path effects degrade the performance of a standard NR based on MWF, especially so when a small amplification G is applied in the hearing aid. This confirms previous observations and simulations results. When the number of sources is less or equal to the number of microphones, the output SNR of the integrated ANC and NR scheme can be derived. The integrated scheme then allows to restore the NR performance and delivers a constant output SNR for any amplification G .

6. REFERENCES

- [1] B. Cornelis, S. Doclo, T. Van den Bogaert, M. Moonen, and J. Wouters. Theoretical analysis of binaural multi-microphone noise reduction techniques. *IEEE Transactions on audio, speech and language processing*, 18(2):342–355, February 2010.
- [2] H. Dillon. *Hearing Aids*. Thieme, 2001.
- [3] 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*, 49(7-8):636–656, 2007.
- [4] S. Elliott and P. Nelson. *active control of sound*. Academic press, Cambridge, 1993.
- [5] S. Elliott and P. Nelson. Active noise control. *Signal Processing Magazine, IEEE*, 10, Issue: 4(1053-5888):12–35, oct 1993.
- [6] L. Griffiths and C. Jim. An alternative approach to linearly constrained adaptive beamforming. *Antennas and Propagation, IEEE Transactions on*, 30:27–34, 1982.
- [7] J. Kiessling. Sounds towards the tympanic membrane. In *8th EFAS Congress*, Heidelberg, June 2007. European Federation of Audiological Societies.
- [8] S. Kuo and D. Morgan. Active noise control: a tutorial review. *Proceedings of the IEEE*, 87, Issue: 6(0018-9219):943–973, Jun 1999.
- [9] D. Morgan and J. Thi. A delayless subband adaptive filter architecture. *Signal Processing, IEEE Transactions on*, 43(8):1819–1830, 1995.
- [10] R. Serizel, M. Moonen, J. Wouters, and S. H. Jensen. Combined active noise control and noise reduction in hearing aids. In *11th International Workshop on Acoustic Echo and Noise Control (IWAENC)*, September 2008.
- [11] R. Serizel, M. Moonen, J. Wouters, and S. H. Jensen. Integrated active noise control and noise reduction in hearing aids. *accepted for publication in IEEE Transactions on Speech Audio and Language*, 2009.