A Robust Null-Steering Beamformer for Acoustic Feedback Cancellation for a Multi-Microphone Earpiece[1]

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Abstract

To reduce acoustic feedback in hearing aids, adaptive filters are commonly used to estimate the feedback contribution in the microphone(s). While theoretically allowing for perfect feedback cancellation, in practice the solution is typically biased due to the closed-loop acoustical system. For an earpiece with multiple integrated microphones and loudspeakers, we propose to use a fixed beamformer to cancel the acoustic feedback. By steering a spatial null in the direction of the hearing aid loudspeaker we show that theoretically perfect feedback cancellation can be achieved. To increase the robustness, the null-steering beamformer is computed based on multiple measured acoustic feedback paths. Experimental results using an earpiece with two microphones in the vent and a third microphone in the concha show that the proposed robust null-steering beamformer substantially and robustly increases the added stable gain while maintaining a high perceptual quality.

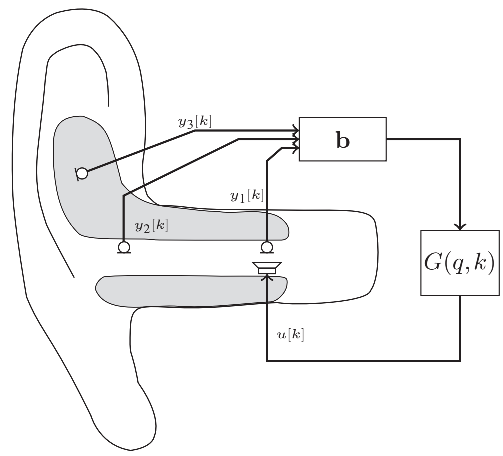

Figure 1: Considered hearing aid setup.

1 Introduction

Due to the acoustic coupling between the hearing aid loudspeaker and microphones(s), acoustic feedback is a common problem limiting the maximum applicable gain in a hearing aid. Most often acoustic feedback is perceived as whistling or howling. In order to increase the maximum gain that can be applied in a hearing aid, robust feedback cancellation strategies are required.

In order to cancel the acoustic feedback, frequently adaptive feedback cancellation schemes are used, i.e., an adaptive filter is used to model the acoustic feedback path between the hearing aid loudspeaker and the microphone(s) [1–6]. Theoretically this allows for perfect cancellation of the acoustic feedback. However, due to the closed-loop system of the hearing aid, the filter adaptation is usually biased, e.g., [7, 8]. Different approaches have been proposed to reduce the bias [3, 8, 9] for single-loudspeaker single-microphone hearing aids. In addition, it has been shown that an improved performance can be achieved by exploiting multiple microphones, e.g., by adaptively removing the contribution of the incoming signal in the filter adaptation [5] or by using a combined multi-microphone feedback cancellation and noise reduction scheme [10, 11].

In this paper, we propose the use of a fixed beamformer, see e.g., [12], to cancel the contribution of the loudspeaker signal in the microphone signals. In particular, we apply this approach to a newly developed earpiece [13] (see Figure 1) with two closely spaced microphones and a loudspeaker in the vent and a third microphone located in the concha. In contrast to conventional behind-the-ear hearing aids, this allows to design a beamformer with a spatial null in the direction of the hearing aid loudspeaker which is located in the vent. Thus the beamformer ideally allows to cancel signals originating from inside of the ear canal and does not impact the incoming signal. In order to improve the robustness against small changes of the position of the earpiece and variations of the sound field, e.g., in the presence of a telephone receiver, we propose to compute the beamformer coefficients based on multiple sets of measured acoustic feedback paths. Experimental results show that the proposed fixed null-steering beamformer enables to reduce the acoustic feedback contribution in a robust way even for unknown acoustic feedback paths, while preserving a high perceptual speech quality in different acoustic conditions.

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2 Acoustic Scenario

Consider a single-loudspeaker multi-microphone hearing aid with M microphones as depicted in Figure 2. The mth microphone signal $y_m[k]$, $m = 1,\ldots,M$, at discrete time k is the sum of the incoming signal $x_m[k]$ and the feedback signal $f_m[k]$, i.e.,

$$\begin{aligned}\mathbf{y}[k] &= \mathbf{x}[k] + \mathbf{f}[k] && (1)\\ &= \mathbf{x}[k] + \mathbf{H}(q,k)u[k], && (2)\end{aligned}$$

with

$$\mathbf{y}[k] = [y_1[k] \quad \ldots \quad y_M[k]]^T, \tag{3}$$

$$\mathbf{x}[k] = [x_1[k] \quad \ldots \quad x_M[k]]^T, \tag{4}$$

$$\mathbf{H}(q,k) = [H_1(q,k) \quad \ldots \quad H_M(q,k)]^T, \tag{5}$$

where $[\cdot]^T$ denotes transpose operation, $u[k]$ is the loudspeaker signal and $H_m(q,k)$ is the acoustic feedback path between the mth microphone and the loudspeaker. We assume that each acoustic feedback path can be modelled as an L_H-dimensional polynomial in the delay element q, i.e.,

$$\begin{aligned}H_m(q,k) &= h_{m,0}[k] + \cdots + h_{m,L_H-1}[k]q^{-L_H+1} && (6)\\ &= \mathbf{h}_m^T[k]\mathbf{q}. && (7)\end{aligned}$$

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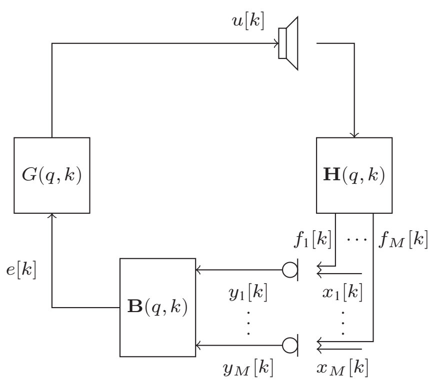

Figure 2: Generic single-loudspeaker multi-microphone hearing aid system.

The L_H-dimensional vector $\mathbf{h}_m[k]$ representing the impulse response of the mth acoustic feedback path is defined as

$$\mathbf{h}_m[k] = [h_{m,0}[k] \quad \dots \quad h_{m,L_H-1}[k]]^T, \tag{8}$$

and $\mathbf{q}$ denotes the vector containing the delay-elements of q of appropriate length.

By applying a (possibly time-varying) filter-and-sum beamformer to the microphone signals, the signal $e[k]$ is obtained, i.e.,

$$e[k] = \mathbf{B}^T(q,k)\mathbf{y}[k], \tag{9}$$

where $\mathbf{B}(q,k)$ is the weighting vector of the beamformer, i.e.,

$$\mathbf{B}(q,k) = [B_1(q,k) \quad \dots \quad B_M(q,k)]^T. \tag{10}$$

The L_B-dimensional beamformer coefficient vector of $B_m(q,k)$ in the mth microphone is defined as

$$\mathbf{b}_m[k] = [b_{m,0}[k] \quad \dots \quad b_{m,L_B-1}[k]]^T, \tag{11}$$

and the ML_B-dimensional stacked vector is defined as

$$\mathbf{b}[k] = \left[\mathbf{b}_1^T[k] \quad \dots \quad \mathbf{b}_M^T[k]\right]^T. \tag{12}$$

The signal $e[k]$ is then processed using the hearing aid forward path $G(q,k)$ resulting in the loudspeaker signal $u[k]$, i.e.,

$$u[k] = G(q,k)e[k]. \tag{13}$$

3 Closed-Loop System Analysis

In the following, we analyse the hearing aid system depicted in Figure 2. By combining (2), (9), and (13) we can rewrite the loudspeaker signal as

$$u[k] = G(q,k)\mathbf{B}^T(q,k)\mathbf{y}[k] \tag{14}$$

$$\begin{aligned} &= G(q,k)\mathbf{B}^T(q,k)\mathbf{x}[k] \\ &\quad + G(q,k)\mathbf{B}^T(q,k)\mathbf{H}(q,k)u[k], \end{aligned} \tag{15}$$

such that

$$u[k] = \underbrace{\frac{G(q,k)\mathbf{B}^T(q,k)}{(1 - G(q,k)\mathbf{B}^T(q,k)\mathbf{H}(q,k))}}_{\mathbf{C}^T(q,k)}\mathbf{x}[k], \tag{16}$$

with $\mathbf{C}(q,k)$ the closed-loop transfer function. From (16) it can be observed that when the beamformer $\mathbf{B}(q,k)$ is able to cancel the contribution of the feedback signals in the microphones perfect feedback cancellation can be achieved, i.e.,

$$\mathbf{B}^T(q,k)\mathbf{H}(q,k) = 0, \tag{17}$$

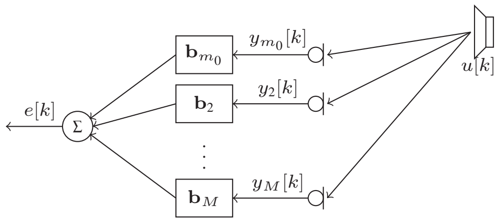

Figure 3: Schematic of the nullsteering beamformer.

with $B_m(q,k) \neq 0$ for at least one $m \in \{1,\dots,M\}$ to avoid the trivial solution. If (17) holds, then (16) reduces to

$$u[k] = G(q,k)\mathbf{B}^T(q,k)\mathbf{x}[k]. \tag{18}$$

Note that although (17) perfectly solves the feedback cancellation problem, from (18) we observe that the null-steering beamformer will also modify the incoming signals $\mathbf{x}[k]$, possibly leading to sound quality degradation.

4 Robust Null-Steering Beamformer

Instead of designing the null-steering beamformer for only one set of (measured) acoustic feedback paths, we aim at increasing the robustness against small changes of the earpiece position by jointly optimizing the null-steering beamformer for a set of I (measured) acoustic feedback paths, $\mathbf{H}^{(i)}(q)$, $i = 1,\dots,I$. This can intuitively be understood as widening the spatial null by using several measurements. The fixed beamformer coefficient vector $\mathbf{b}$ can hence be computed by minimizing the mean least-squares cost function

$$\boxed{J_{LS}(\mathbf{b}) = \sum_{i=1}^{I} \|(\mathbf{H}^{(i)})^T\mathbf{b}\|_2^2} \tag{19}$$

where $\mathbf{H}^{(i)}$ is the $ML_B \times (L_B + L_H - 1)$-dimensional matrix of concatenated convolution matrices of the acoustic feedback paths from the ith set, $i = 1,\dots,I$, i.e.,

$$(\mathbf{H}^{(i)})^T = \left[(\mathbf{H}_1^{(i)})^T \quad \dots \quad (\mathbf{H}_M^{(i)})^T\right] \tag{20}$$

where $\mathbf{H}_m^{(i)}$ is the $L_B \times (L_B + L_H - 1)$-dimensional convolution matrix of $\mathbf{h}_m^{(i)}$.

In order to prevent the trivial solution of $\mathbf{b} = \mathbf{0}$, we constrain the beamformer coefficients in a reference microphone m_0 to correspond to a delay of L_d samples, i.e.,

$$\mathbf{b}_{m_0} = [\underbrace{\begin{matrix} 0 & \dots & 0 \end{matrix}}_{L_d} \quad 1 \quad 0 \quad \dots \quad 0\ ]^T. \tag{21}$$

The constrained LS problem in (19) and (21) can be reformulated as

$$\boxed{\tilde{J}_{LS}(\tilde{\mathbf{b}}) = \sum_{i=1}^{I} \|\mathbf{h}_{m_0}^{(i)} + \sum_{\substack{m=1\\m\neq m_0}}^{M} (\mathbf{H}_m^{(i)})^T\mathbf{b}_m\|_2^2} \tag{22}$$

with the closed-form solution

$$\tilde{\mathbf{b}}_{LS} = (\tilde{\mathbf{H}}\tilde{\mathbf{H}}^T)^{-1}\tilde{\mathbf{H}}\tilde{\mathbf{h}}_{m_0}, \tag{23}$$

where the solution $\tilde{\mathbf{b}}_{LS}$ contains the beamformer coefficients for microphones $m = 1,\dots,M$, $m \neq m_0$, $\tilde{\mathbf{H}}$ is the $(M-1)L_B \times (L_B + L_H - 1)I$-dimensional matrix of stacked and concatenated convolution matrices $\mathbf{H}_m^{(i)}$, $m = 1,\dots,M, m \neq m_0$, $i = 1,\dots,I$ and $\tilde{\mathbf{h}}_{m_0}$ is the $(L_B + L_H - 1)I$-dimensional vector of stacked vectors $\mathbf{h}_{m_0}^{(i)}$, $i = 1,\dots,I$.

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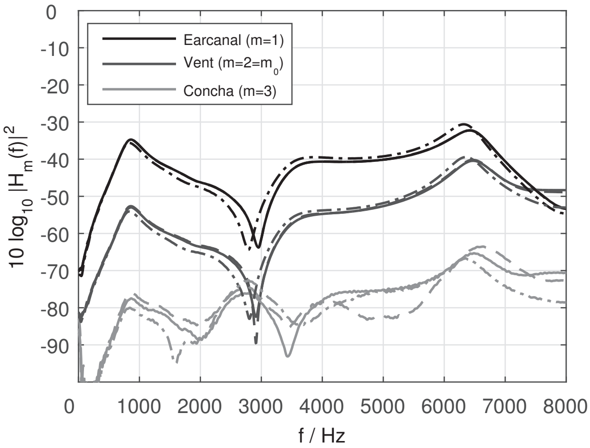

Figure 4: Amplitude response of the measured acoustic feedback paths. Continuous lines show exemplary feedback paths without any obstruction (i.e., in free-field), dashed dotted lines show an exemplary responses after repositioning of the earpiece, and dashed lines show the acoustic feedback paths in the presence of a telephone receiver.

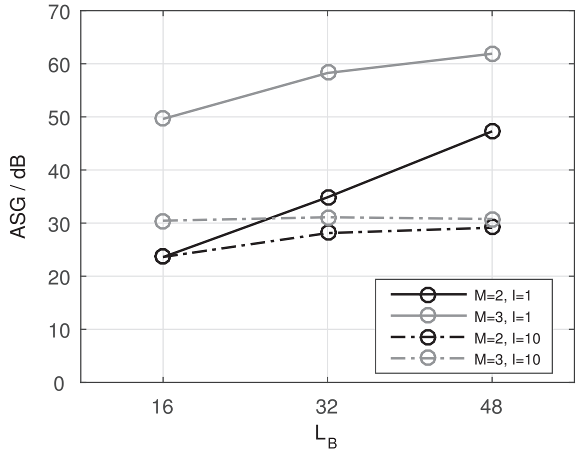

Figure 5: Average ASG for different numbers of microphones and different sets of acoustic feedback paths as a function of the beamformer filter length L_B for the optimal performance.

Table 1: Average PESQ score for different numbers of microphones and different sets of acoustic feedback paths as a function of the beamformer filter length L_B.

		$L_B = 16$	$L_B = 32$	$L_B = 48$
$M = 2$	$I = 1$	4.44	4.44	4.42
	$I = 10$	4.44	4.44	4.42
$M = 3$	$I = 1$	4.32	4.26	4.28
	$I = 10$	4.30	4.31	4.27

5 Experimental Evaluation

In this section the performance of the proposed average null-steering beamformer is evaluated when using $M = 2$ or $M = 3$ microphones. In particular we consider the ability to cancel the acoustic feedback in different acoustic scenarios as well as the resulting distortion of the incoming signal $\mathbf{x}[k]$.

5.1 Setup and Performance Measures

Acoustic feedback paths were measured for the three-microphone earpiece depicted in Figure 1 on a dummy head with adjustable ear canals [14]. The impulse responses were sampled at $f_s = 16$ kHz and truncated to length $L_H = 100$. Measurements were performed in an acoustically treated chamber. Figure 4 shows exemplary amplitude responses of the measured acoustic feedback paths for the three different microphones and for two different acoustic conditions. In total 20 different sets of acoustic feedback paths were measured, i.e., the earpiece was repositioned on the dummy head 10 times and for each repositioning feedback paths were measured in both free-field, i.e., without obstruction, and with a telephone receiver in close distance to the ear. The forward path of the hearing aid was of the set to $G(q,k) = q^{-96}10^{45/20}$, corresponding to a delay of 6 ms and a broadband amplification of 45 dB. For all experiments $L_d = L_B/2$ was chosen and the reference microphone $m_0 = 2$, i.e., the microphone located at the outer phase of the vent, was chosen which contains the most relevant perceptual cues for an external sound source. For $M = 2$ microphones $m = 1,2$ were used, while for $M = 3$ microphones $m = 1,2,3$ were used.

We evaluated the feedback cancellation performance of the beamformer using the added stable gain (ASG) [8] and the perceptual quality using the perceptual quality of speech (PESQ) measure [15]. The ASG for the considered hearing aid setup is computed as [8]

$$ASG = 20\log_{10} \frac{1}{\max\limits_{f} \left| \sum\limits_{m=1}^{M} H_m(f) B_m(f) \right|} - MSG_{m_0}, \quad (24)$$

where f is the frequency and MSG_{m_0} is the maximum stable gain in the reference microphone m_0 without applying the beamformer, i.e.,

$$MSG_{m_0} = 20\log_{10} \frac{1}{\max_f |H_{m_0}(f)|}. \quad (25)$$

The reference signal for the PESQ measure was the incoming signal $x_{m_0}[k]$ in the reference microphone, while the test signal was the error signal $e[k]$ after applying the beamformer. As speech signal we concatenated 26 sentences spoken by 4 different speakers from the TIMIT database [16] resulting in an 80 s long signal. The distance between the external source and the dummy head was 1.2 m.

5.2 Experiment 1: Optimal performance

In the first experiment we evaluate the optimal performance of the propose null-steering beamformer using feedback paths measured in free-field. The beamformer coefficients vector $\mathbf{b}$ was computed using (23), either using only one set of measured feedback paths ($I = 1$), resulting in 10 different beamformers, or using all available sets of measured feedback paths ($I = 10$), resulting in 1 beamformer. The average performance measures (ASG, PESQ) were computed by averaging these measures over all available sets of feedback paths, where for $I = 1$ only the feedback paths included in the optimization were used for evaluation. Figure 5 shows the average ASG as a function of the beamformer filter length L_B. As expected, using $I = 1$ leads to a larger average ASG compared to using $I = 10$. Similarly, using $M = 3$ microphones leads to a larger average ASG compared to using $M = 2$ microphones. Table 1 shows the average PESQ scores, indicating that the perceptual quality is practically not influenced since the PESQ scores are larger than 4.27.

5.3 Experiment 2: Internal sound field variations

In the second experiment, we evaluate the robustness of the proposed null-steering beamformer against internal sound field variations, as it has been shown that only small changes of the hearing aid position may alter the acoustic feedback path [17]. We

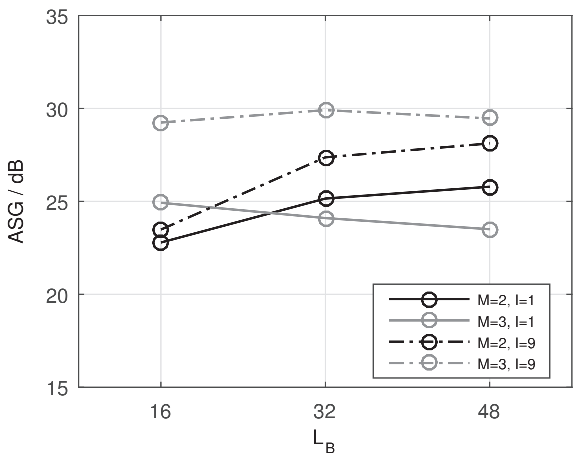

Figure 6: Average ASG for different numbers of microphones and different sets of acoustic feedback paths as a function of the beamformer filter length L_B for internal sound field variations.

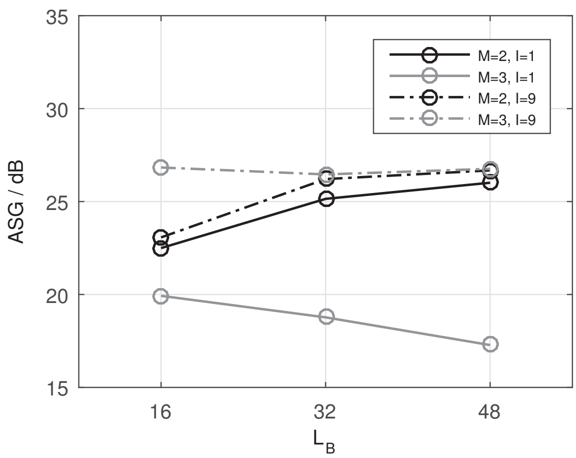

Figure 7: Average ASG for different numbers of microphones and different sets of acoustic feedback paths as a function of the beamformer filter length L_B for external sound field variations.

Table 2: Average PESQ score for different numbers of microphones and different sets of acoustic feedback paths as a function of the beamformer filter length L_B for internal sound field variations.

		$L_B = 16$	$L_B = 32$	$L_B = 48$
$M = 2$	$I = 1$	4.44	4.44	4.42
	$I = 9$	4.44	4.44	4.42
$M = 3$	$I = 1$	4.30	4.31	4.27
	$I = 9$	4.32	4.27	4.27

Table 3: Average PESQ score for different numbers of microphones and different sets of acoustic feedback paths as a function of the beamformer filter length L_B for external sound field variations.

		$L_B = 16$	$L_B = 32$	$L_B = 48$
$M = 2$	$I = 1$	4.43	4.43	4.41
	$I = 9$	4.43	4.43	4.41
$M = 3$	$I = 1$	4.30	4.32	4.27
	$I = 9$	4.33	4.28	4.26

consider two different sets for computing the beamformer: a) $I = 1$, where the performance measures are computed by averaging the performance over (the remaining) nine acoustic feedback path measurements, and b) $I = 9$, where the evaluation is performed using the tenth free-field measurement that was not included in the optimization, i.e., using a leave-one-out cross-validation approach. Note that for both cases ten different beamformers are computed. Figure 6 shows the average ASG, showing that the proposed robust null-steering beamformer leads to an increased average performance compared to using only a single set of acoustic feedback paths to compute the beamformer. Using $I = 9$ compared to using $I = 1$ leads to an improvement of about 2-3 dB for $M = 2$ microphones and about 5-6 dB for $M = 3$ microphones, indicating an increased robustness. This means that even when repositioning the earpiece, the proposed robust null-steering beamformer yields an average ASG of about 30 dB. Table 2 shows the average PESQ scores indicating that there is practically no influence of the beamformer on the perceived speech quality.

5.4 Experiment 3: External sound field variations

In the third experiment we evaluate the performance of the proposed null-steering beamformer to unknown external sound field variations. Therefore we use the beamformers computed in Experiment 2 using the free-field feedback path measurements. However, instead of using the free-field feedback path measurement for evaluation, here we use the acoustic feedback paths measured with a telephone receiver in close distance. Thus, this condition includes both internal and external sound field variations. Figure 7 shows the average ASG, showing that the proposed robust beamformer leads to an average ASG of 23-26 dB for $M = 2$ and 26 dB for $M = 3$, while using only a single set of acoustic feedback paths leads to lower average ASGs. Table 3 shows the corresponding average PESQ scores, indicating that again the perceived quality is practically not reduced. These results indicate the advantage of computing the beamformer based on multiple sets of acoustic feedback paths to achieve robustness to unknown acoustic feedback paths, both for internal and external sound field variations.

6 Conclusions

In this paper we proposed a fixed robust beamformer to perform acoustic feedback cancellation in an earpiece with multiple integrated microphones by steering a spatial null in the direction of the hearing aid loudspeaker. We formulated the estimation of the beamformer coefficients as a least-squares optimization problem, where we constrain the beamformer coefficients in a reference microphone to a delay and compute the beamformer coefficients based on multiple set of measured acoustic feedback paths. Experimental results show that the proposed robust beamformer leads to a larger ASG compared to using only a single set acoustic feedback paths to compute the beamformer coefficients, while maintaining a good perceptual quality, even when placing a telephone close to the ear or repositioning the earpiece . In conclusion, by using the proposed robust null-steering beamformer the ASG is improved in a robust way by more than 20 dB while not compromising the speech quality.

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