

# NULL-STEERING BEAMFORMER FOR ACOUSTIC FEEDBACK CANCELLATION IN A MULTI-MICROPHONE EARPIECE OPTIMIZING THE MAXIMUM STABLE GAIN

Henning Schepker<sup>1</sup>, Linh T. T. Tran<sup>2</sup>, Sven Nordholm<sup>2</sup>, and Simon Doclo<sup>1</sup>

<sup>1</sup> Signal Processing Group, Department of Medical Physics and Acoustics and Cluster of Excellence "Hearing4All", University of Oldenburg, Oldenburg, Germany  
{henning.schepker, simon.doclo}@uni-oldenburg.de

<sup>2</sup> Faculty of Science and Engineering, Curtin University, Perth, Australia  
t.tran57@postgrad.curtin.edu.au, S.Nordholm@curtin.edu.au

## ABSTRACT

Commonly adaptive filters are used to reduce the acoustic feedback in hearing aids. While theoretically allowing for perfect cancellation of the feedback signal, in practice the adaptive filter solution is typically biased due to the closed-loop hearing aid system. In contrast to conventional behind-the-ear hearing aids, in this paper we consider an earpiece with multiple integrated microphones. For such an earpiece it has previously been proposed to use a fixed beamformer to reduce the acoustic feedback in the microphones which has been designed to minimize a least-squares cost function. In this paper we propose to design the beamformer by minimizing a min-max cost function which directly maximizes the maximum stable gain of the earpiece. Furthermore, we propose a robust extension of the min-max cost function maximizing the worst-case maximum stable gain over a set of acoustic feedback paths. Experimental results using measured acoustic feedback paths show that the feedback cancellation performance of the fixed beamformer can be considerably improved by minimizing the proposed min-max optimization problem, while maintaining a high perceptual quality of the incoming signal.

**Index Terms**— acoustic feedback cancellation, null-steering beamformer, min-max optimization, hearing aids

## 1. INTRODUCTION

Acoustic feedback is a common problem in hearing aids which occurs due to acoustic coupling between the hearing aid loudspeaker and the microphone(s), thus limiting the maximum applicable gain. Often the acoustic feedback degrades the quality and is perceived as whistling or howling. Therefore, in order to increase the maximum gain and increase the sound quality in a hearing aid, robust feedback cancellation strategies are needed.

Frequently adaptive feedback cancellation schemes are used to cancel the acoustic feedback, which use an adaptive filter to model the acoustic feedback path(s) between the hearing aid loudspeaker and the microphone(s) [1, 2, 3, 4, 5, 6]. Theoretically these approaches allow for perfect cancellation of the feedback signal. However, due to the closed-loop system of the hearing aid, the filter adaptation is usually biased [7, 8]. In order to reduce this bias several solutions have been proposed for single-loudspeaker single-microphone hearing aids [3, 8, 9]. Furthermore, several approaches to improve the feedback cancellation performance using multiple microphones have been proposed, e.g., by adaptively removing the

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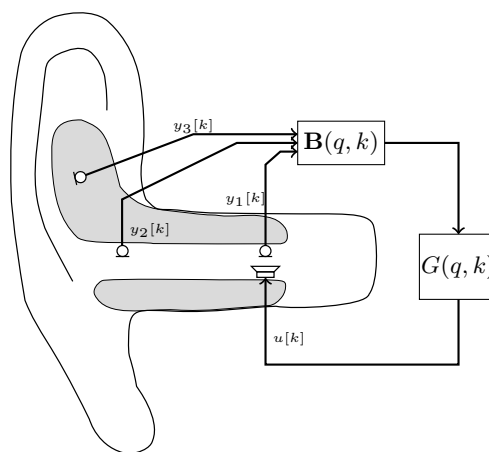
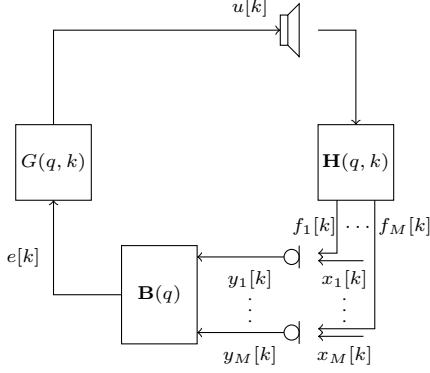


Fig. 1. Considered hearing aid setup.

contribution of the incoming signal in the filter adaptation [5], using a fixed null-steering beamformer [10, 11] or by using a combined multi-microphone feedback cancellation and noise reduction scheme [12, 13].

In this paper we consider the design of a fixed null-steering beamformer to cancel the contribution of the loudspeaker signal in the microphone(s) of a newly developed earpiece [15]. Figure 1 depicts the design of the earpiece with two closely spaced microphones and a loudspeaker in the vent and a third microphone located in the concha. This physical design allows to design a beamformer with a spatial null in the direction of the hearing aid loudspeaker located in the vent. Hence, ideally the beamformer provides cancellation of signals originating from the inside of the ear canal while not (largely) impacting the incoming signal.

While in [10, 11] we proposed to design the null-steering beamformer using a least-squares (LS) cost function, in this paper we propose to compute the null-steering beamformer coefficients by optimizing a min-max cost function. Similar to the min-max design of filters for adaptive feedback cancellation [16, 17] we show that the min-max design of a null-steering beamformer for acoustic feedback cancellation corresponds to a direct optimization of the maximum stable gain. Furthermore, we propose to increase the robustness of the min-max null-steering beamformer by computing those filters that optimize the worst-case performance over a set of different measured acoustic feedback paths. Results show that the added stable gain can be considerably improved by this direct optimization of the



**Fig. 2.** Generic single-loudspeaker multi-microphone hearing aid system.

maximum stable gain, while maintaining a high perceptual speech quality of the incoming signal.

## 2. ACOUSTIC SETUP

Consider the single-loudspeaker multi-microphone scenario with  $M$  microphones depicted in Figure 2. The  $m$ th microphone signal  $y_m[k]$  at discrete time  $k$  is the sum of the incoming signal  $x_m[k]$  and the feedback signal  $f_m[k] = H_m(q, k)u[k]$ , i.e.,

$$y_m[k] = x_m[k] + f_m[k], \quad (1)$$

where  $u[k]$  is the loudspeaker signal and  $H_m(q, k)$  is the acoustic feedback path between the  $m$ th microphone and the loudspeaker. We assume that the  $m$ th acoustic feedback path can be modeled as an  $L_H$ -dimensional polynomial in the delay element  $q$ , i.e.,

$$H_m(q, k) = h_{m,0}[k] + \dots + h_{m,L_H-1}q^{-L_H+1}. \quad (2)$$

The output  $e[k]$  of the fixed beamformer  $B(q)$  is computed as

$$e[k] = \sum_{m=1}^M B_m(q)y_m[k], \quad (3)$$

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

$$u[k] = G(q, k)e[k]. \quad (4)$$

By combining equations (1), (3), and (4), it can be shown that the closed-loop transfer function of the considered hearing aid system in Figure 2 is [10]

$$\mathcal{C}^T(q, k) = \frac{G(q, k)\mathbf{B}^T(q)}{1 - G(q, k)\mathbf{B}^T(q)\mathbf{H}(q, k)}, \quad (5)$$

with the stacked vectors

$$\mathbf{B}(q) = [B_1(q) \quad \dots \quad B_M(q)]^T, \quad (6)$$

$$\mathbf{H}(q, k) = [H_1(q, k) \quad \dots \quad H_M(q, k)]^T. \quad (7)$$

## 3. MAXIMUM STABLE GAIN OF HEARING AID SYSTEM

The considered hearing aid setup is stable if the denominator of the closed-loop transfer function in (5) is larger than 0 for all frequencies  $\omega$ , i.e.,  $|G(\omega)\mathbf{B}^H(\omega)\mathbf{H}(\omega)| < 1$ . Assuming a broadband forward path gain  $G(\omega) = G$  as in [18] the *maximum stable gain*  $\mathcal{M}_i$  for the  $i$ th measurement of the acoustic feedback paths,  $i = 1, \dots, I$  is given by

$$\mathcal{M}_i = \frac{1}{\max_{\omega} |\mathbf{H}_i^H(\omega)\mathbf{B}(\omega)|^2}, \quad (8)$$

where  $\mathbf{H}_i(\omega)$  is the stacked vector of the frequency responses at frequency  $\omega$  of each microphone of the  $i$ th measurement and  $\mathbf{B}(\omega)$  is the stacked vector of the frequency responses of the beamformer in each microphone, i.e.,

$$\mathbf{H}_i(\omega) = [H_{1,i}(\omega) \quad \dots \quad H_{M,i}(\omega)]^H, \quad (9)$$

$$\mathbf{B}(\omega) = [b_1(\omega) \quad \dots \quad b_M(\omega)]^H. \quad (10)$$

Furthermore, we defined the *overall maximum stable gain* for a considered set of  $I$  measurements as

$$\mathcal{M} = \min_i \mathcal{M}_i, \quad i = 1, \dots, I. \quad (11)$$

## 4. MIN-MAX DESIGN

In this section we will first review the LS null-steering design proposed in [10] and propose two new alternatives to design the null-steering beamformer that aim at maximizing the maximum stable gain in (8) and maximizing the overall maximum stable gain in (11).

In [10] it was proposed to compute the optimal beamformer coefficients for a single set of measured acoustic feedback paths by minimizing the following LS cost function

$$\begin{aligned} \min_{\mathbf{b}} \quad & \|\tilde{\mathbf{H}}^T \mathbf{b}\|_2^2 \\ \text{subject to} \quad & \mathbf{b}_{m_0} = \underbrace{[0 \quad \dots \quad 0]}_{L_d} \quad 1 \quad 0 \quad \dots \quad 0 \quad ]^T, \end{aligned} \quad (12)$$

where  $\tilde{\mathbf{H}}$  is the  $(ML_B) \times (L_B + L_H - 1)$ -dimensional matrix of concatenated  $(L_B) \times (L_B + L_H - 1)$ -dimensional convolution matrices  $\tilde{\mathbf{H}}_m$  and  $\mathbf{b}$  is the  $ML_B$ -dimensional vector of the concatenated  $L_B$ -dimensional beamformer coefficient vectors  $\mathbf{b}_m$ , i.e.,

$$\tilde{\mathbf{H}}^T = [\tilde{\mathbf{H}}_1^T \quad \dots \quad \tilde{\mathbf{H}}_M^T], \quad (13)$$

$$\mathbf{b} = [\mathbf{b}_1 \quad \dots \quad \mathbf{b}_M]^T. \quad (14)$$

The constraint in (12) is added to avoid the trivial solution of  $\mathbf{b} = \mathbf{0}$  and essentially selects a reference microphone  $m = m_0$  in which the feedback is cancelled by using the remaining microphones, with  $L_d$  a delay to account for acausalities.

While optimizing the cost function in (12) may lead to reasonable performance in terms of the maximum stable gain, it is not directly related to the maximum stable gain. Therefore, we propose to design the null-steering beamformer such that it directly optimizes the *maximum stable gain* defined in (8). Maximizing the maximum stable gain corresponds to minimizing the denominator in (8), which can be formulated as the following min-max optimization problem

$$\begin{aligned} \min_{\mathbf{b}} \max_{\omega} \quad & |\mathbf{H}^H(\omega)\mathbf{B}(\omega)|^2 \\ \text{s.t.} \quad & \mathbf{b}_{m_0} = \underbrace{[0 \quad \dots \quad 0]}_{L_d} \quad 1 \quad 0 \quad \dots \quad 0 \quad ]^T \end{aligned} \quad (15)$$

where  $\mathbf{H}(\omega)$  and  $\mathbf{B}(\omega)$  are defined in (9) and (10). The min-max optimization problem in (15) can be approximated as the following linear programming problem [19]

$$\begin{aligned} \min_{\mathbf{b}} \quad & t & (16a) \\ \text{s.t.} \quad & t \geq 0 & (16b) \\ & |\mathbf{c}^T(\omega)\mathbf{H}^T\mathbf{b}| \leq t \quad \forall\omega & (16c) \\ & |\mathbf{s}^T(\omega)\mathbf{H}^T\mathbf{b}| \leq t \quad \forall\omega & (16d) \\ & \mathbf{b}_{m_0} = \underbrace{[0 \ \dots \ 0]}_{L_d} [1 \ 0 \ \dots \ 0]^T & (16e) \end{aligned}$$

where  $\mathbf{c}(\omega)$  and  $\mathbf{s}(\omega)$  are defined as

$$\mathbf{c}(\omega) = [1 \ \cos\omega \ \dots \ \cos(L_B + L_H - 1)\omega]^T, \quad (17)$$

$$\mathbf{s}(\omega) = [0 \ \sin\omega \ \dots \ \sin(L_B + L_H - 1)\omega]^T, \quad (18)$$

essentially computing the real part and imaginary part of the frequency response in (16c) and (16d), respectively. The frequency response is then evaluated over a dense grid of  $Q$  frequencies. The linear programming problem can be efficiently solved using existing convex optimization toolboxes like CVX [20, 21].

In order to improve the robustness we further propose to maximize the *overall maximum stable gain* as defined in (11), which can be formulated as the following min-max optimization problem

$$\begin{aligned} \min_{\mathbf{b}} \max_{\omega, i} \quad & |\mathbf{H}_i^H(\omega)\mathbf{B}(\omega)|^2 \\ \text{s.t.} \quad & \mathbf{b}_{m_0} = \underbrace{[0 \ \dots \ 0]}_{L_d} [1 \ 0 \ \dots \ 0]^T \end{aligned} \quad (19)$$

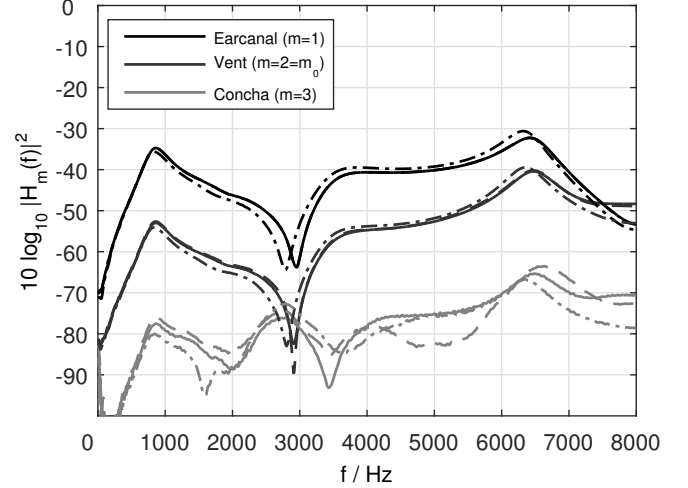
Note that the optimization problems in (15) and (19) are equivalent when only a single ( $I = 1$ ) set of measurements is considered. Similar to the min-max optimization problem in (15) the min-max optimization problem in (19) can be approximated as a linear programming by optimizing over all  $\omega$  and  $i, i = 1, \dots, I$ .

## 5. EXPERIMENTAL EVALUATION

In this section the performance of the proposed min-max null-steering beamformer designs are evaluated when using  $M = 2$  (i.e.,  $m = 1, 2$ ) or  $M = 3$  (i.e.,  $m = 1, 2, 3$ ) microphones and compared to the LS null-steering beamformer proposed in [10]. We consider both the ability to cancel the acoustic feedback in different acoustic scenarios as well as the resulting distortion of the incoming signal  $\mathbf{x}[k]$  due to the beamformer.

### 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 [22]. 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 3 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



**Fig. 3.** 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.

broadband amplification of 45 dB. For all experiments the reference microphone  $m_0 = 2$ , i.e., the microphone located at the outer phase of the vent, was chosen as it provides a natural position for sound pickup likely including all relevant perceptual cues,  $L_d = L_B/2$  and  $Q = 2048$ .

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 [23]. The  $ASG_i$  of the  $i$ th set of measurements for the considered hearing aid setup is computed as [8]

$$ASG_i = 20 \log_{10} \mathcal{M}_i - 20 \log_{10} \frac{1}{\max_f |H_{i,m_0}(f)|}, \quad (20)$$

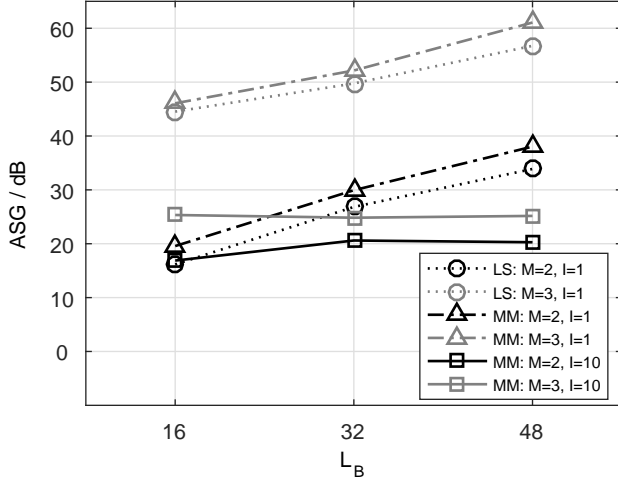
where the second term is the maximum stable gain in the reference microphone  $m_0$  without applying the beamformer. Furthermore, we define the overall ASG as

$$ASG = \min_i ASG_i. \quad (21)$$

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 a speech signal we used sentences from the TIMIT database [24], where we concatenated 26 sentences spoken by 4 different speakers 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 proposed beamformer design methods using the overall ASG and the PESQ measure. The beamformer coefficient vector was computed using the acoustic feedback paths measured in free-field, resulting in 10 different beamformers for the optimization methods in (12) and (15), or using all available measurements, resulting in a single beamformer for the optimization method in (19). Figure 4 depicts the overall ASG for the proposed min-max cost functions optimizing the MSG in (15) and (19) as well as the LS cost function



**Fig. 4.** Overall ASG for different numbers of microphones and beamformer design methods as a function of the beamformer length  $L_B$ .

**Table 1.** Worst-case PESQ scores for different numbers of microphones and cost functions as a function of the beamformer filter length  $L_B$ .

		$L_B = 16$	$L_B = 32$	$L_B = 48$
$M = 2$	LS in (12)	4.44	4.43	4.40
	MM in (15)	4.43	4.43	4.40
	MM in (19)	4.42	4.43	4.40
$M = 3$	LS in (12)	4.27	4.27	4.14
	MM in (15)	4.20	4.15	4.08
	MM in (19)	4.25	4.23	4.19

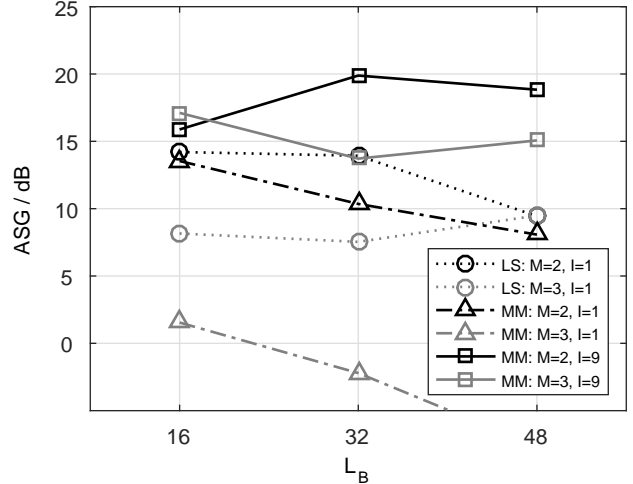
proposed in [10]. As can be seen all design methods lead to large ASGs. Moreover, it can be observed that for  $I = 1$  the min-max optimization outperform the LS optimization by approximately 4 dB. As expected the performance is reduced when using  $I = 10$  compared to  $I = 1$ . Furthermore, using  $M = 3$  leads to an increase in performance compared to using  $M = 2$ .

Table 1 shows the worst-case PESQ scores, i.e., the minimum PESQ score obtained for the different measurements. As can be seen all PESQ scores are above a value of 4.0 indicating that the quality of the processed incoming speech signal is not significantly influenced by the proposed null-steering beamformer design methods.

### 5.3. Experiment 2: Sound field variations

In the second experiment we evaluate the robustness of the proposed null-steering beamformer design methods against the combined change of internal and external sound field variations, as it has been shown that these may significantly alter the acoustic feedback paths [25]. We consider two different sets for computing the beamformer coefficient vectors: a)  $I = 1$ , for the design methods in (12) and (15) where the performance measures are computed as the minimum performance for the remaining nine acoustic feedback paths measurements, and b)  $I = 9$ , where the evaluation is performed using the tenth acoustic feedback path measurement that was not included in the optimization, i.e., a leave-one out cross validation approach. Note that the beamformer coefficients were computed using the free-field measurements and evaluations were performed in the presence of a telephone receiver.

Figure 5 shows the overall ASG, indicating that the performance is drastically reduced when only a single set of measurements ( $I = 1$ ) is used to compute the beamformer coefficients, while using



**Fig. 5.** Overall ASG as a function of the beamformer filter length  $L_B$  when the beamformer coefficients are optimized using the free-field measurements and evaluations are performed using the telephone measurements.

**Table 2.** Worst-case PESQ scores for different numbers of microphones and cost functions as a function of the beamformer filter length  $L_B$  when the beamformer coefficients are optimized using the free-field measurements and evaluations are performed using the telephone measurements. US indicates an unstable system.

		$L_B = 16$	$L_B = 32$	$L_B = 48$
$M = 2$	LS in (12)	4.42	4.41	4.39
	MM in (15)	4.41	4.41	4.39
	MM in (19)	4.41	4.41	4.40
$M = 3$	LS in (12)	US	4.27	4.14
	MM in (15)	US	US	US
	MM in (19)	4.20	4.20	3.98

$I = 9$  leads to a robust overall ASG of at least 15 dB. Furthermore, it can be observed that using  $M = 3$  is generally less robust to changes in the acoustic sound field than using  $M = 2$ . This is in contrast to Experiment 1 where using the third microphone led to an increase in performance. Table 2 shows the worst-case PESQ scores indicating a high perceptual speech quality with PESQ scores larger than about 4, when the system is stable. These results indicate the benefit of the proposed design method when robustly optimizing the overall maximum stable gain.

## 6. CONCLUSION

In this paper we proposed a fixed 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 proposed to directly optimize the maximum stable gain of the hearing aid, which leads to a min-max optimization problem. We proposed two different approaches, where first we only use a single measurement to compute the null-steering beamformer and second we optimize over set of multiple measurements to increase the robustness when optimizing the overall maximum stable gain. Experimental results using measured acoustic feedback paths show that the proposed (robust) optimization approaches lead to a large added stable gain, while maintaining a good perceptual quality. The proposed robust approach performs well even if a telephone is close to the ear or the earpiece is repositioned.

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