ACTIVE FEEDBACK SUPPRESSION FOR HEARING DEVICES EXPLOITING MULTIPLE LOUDSPEAKERS

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ABSTRACT

In hearing devices, acoustic feedback frequently occurs due to the coupling between the hearing device loudspeaker(s) and microphone(s). In order to remove the feedback component from the microphone(s), adaptive filters are commonly used. While many hearing devices contain only a single loudspeaker, in this paper we consider a hearing device with multiple loudspeakers in the vent of a custom earpiece. We exploit this availability by pre-processing the loudspeaker signals such that they interfere destructively at the hearing device microphone while the signal at the eardrum is preserved. More specifically, we design a spatial pre-processor that aims at maximizing the maximum stable gain while limiting the distortions of the desired signal at the eardrum. Experimental results using measured impulse responses from a custom hearing device with two loudspeakers show that the proposed approach yields a robust reduction of the acoustic feedback while preserving the desired signal at the eardrum.

Index Terms— feedback suppression, multiple loudspeakers, hearing devices, maximum stable gain

1. INTRODUCTION

Acoustic feedback due to the coupling between the loudspeaker(s) and the microphone(s) is a common problem that limits the maximum applicable gain and the sound quality in assistive listening devices, e.g., hearing aids. The sound quality degradations due to acoustic feedback are often perceived as whistling or howling. In order to improve the sound quality and increase the maximum applicable gain, robust feedback suppression algorithms are required.

Different techniques for acoustic feedback suppression have been proposed that include non-linear modifications of the loudspeaker signal, using adaptive filtering techniques or using multi-microphone feedback reduction algorithms (see also [1, 2]). While non-linear modifications of the loudspeaker signal generally introduce some audible distortions, adaptive filtering techniques that model the acoustic feedback path(s), in theory, allow for perfect feedback suppression without sound quality degradations. Although in practice the adaptive filter may converge to a biased solution due to the closed-loop system of the hearing device [3, 4], several approaches have been proposed to reduce this bias, e.g., by using prewhitening filters or additional non-linear modifications [5, 6, 7, 8, 9, 10, 11]. The availability of multiple hearing device microphones either enables to further improve the performance of adaptive filtering techniques [12, 13], allows to combined noise reduction and feedback cancellation [14, 15] or enables the usage of a (fixed) beamformer steering a spatial null towards the location of the hearing aid loudspeaker [16].

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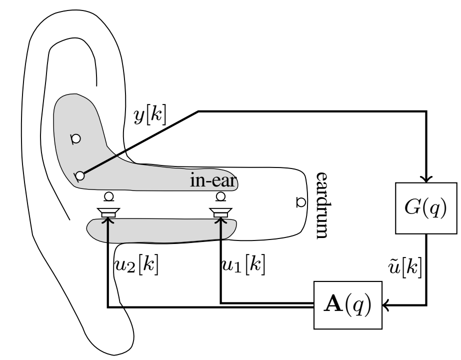


Figure 1: Schematic overview of considered single-microphone multi-loudspeaker hearing device setup.

All of the aforementioned approaches assume that only a single loudspeaker is available in the hearing device. In this paper we consider a hearing device with multiple loudspeakers in the vent as depicted in Figure 1. The availability of multiple loudspeakers allows to steer a spatial null towards the location of the hearing aid microphone picking up the sound signal, hence, *actively* suppressing the feedback component in the microphone signal. It should be noted that unlike typical active noise control strategies [17, 18], the proposed approach does not require a low processing delay. We propose to design the feedback suppression filter by maximizing the maximum stable gain of the hearing device while limiting the distortions of the signal arriving at the eardrum. The resulting optimization problem is formulated as a quadratically constrained quadratic programming problem. In order to improve the robustness to possible variations of the acoustic feedback paths, e.g., due to a telephone receiver close to the ear, we incorporate multiple measurements of the acoustic feedback paths into the optimization problem. Experimental results using measured impulse responses from a custom hearing device prototype with two loudspeakers [19] show that the proposed multi-loudspeaker feedback suppression approach allows to robustly reduce feedback by approximately 4–5 dB without significantly distorting the signal at the eardrum.

2. ACOUSTIC SCENARIO AND NOTATION

Consider the hearing device setup depicted in Figure 2 with N loudspeakers and a single microphone. For simplicity we assume that all acoustic transfer functions are linear and time-invariant and can be modeled as polynomials in the variable q [20]. The microphone signal

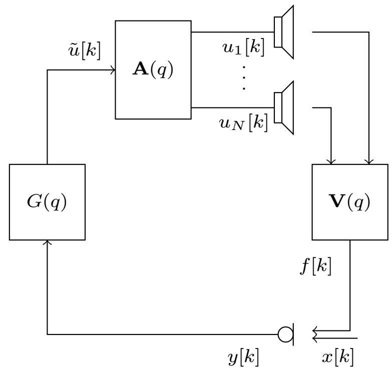


Figure 2: Considered multi-loudspeaker single-microphone hearing device setup.

$y[k]$ at discrete time k is the sum of the incoming signal $x[k]$ and the feedback contributions from all loudspeakers, i.e.,

$$y[k] = x[k] + \underbrace{\mathbf{V}^T(q)\mathbf{u}[k]}_{f[k]}, \tag{1}$$

with the N-dimensional vectors

$$\mathbf{V}(q) = \begin{bmatrix} V_1(q) & V_2(q) & \dots & V_N(q) \end{bmatrix}^T, \tag{2}$$

$$\mathbf{u}[k] = \begin{bmatrix} u_1[k] & u_2[k] & \dots & u_N[k] \end{bmatrix}^T, \tag{3}$$

where $[\cdot]^T$ denotes transpose operation, $u_n[k]$ is the n-th loudspeaker signal and $V_n(q)$ is the acoustic feedback paths between the n-th loudspeaker and the microphone. The acoustic feedback paths are modeled as L_V-dimensional polynomials, i.e.,

$$V_n(q) = \mathbf{v}_n^T \mathbf{q} = \sum_{i=0}^{L_V-1} v_{n,i} q^{-i}, \tag{4}$$

where $\mathbf{v}_n$ denotes the L_V-dimensional vector of the impulse response of the n-th acoustic feedback path and $\mathbf{q}$ denotes the vector containing the delay elements of q of appropriate length. The microphone signal is then processed by the hearing aid forward path $G(q)$, yielding the intermediate signal $\tilde{u}[k]$, i.e.,

$$\tilde{u}[k] = G(q)y[k]. \tag{5}$$

This intermediate signal is then processed by applying a spatial pre-processor $\mathbf{A}(q)$ to generate the loudspeaker signals, i.e.,

$$\mathbf{u}[k] = \mathbf{A}(q)\tilde{u}[k], \tag{6}$$

where $\mathbf{A}(q)$ is the N-dimensional vector of the spatial pre-processor weighting functions, i.e.,

$$\mathbf{A}(q) = \begin{bmatrix} A_1(q) & A_2(q) & \dots & A_N(q) \end{bmatrix}^T. \tag{7}$$

The L_A-dimensional coefficient vector $\mathbf{a}_n$ of the impulse response of $A_n(q)$ for the n-th loudspeaker is given by

$$\mathbf{a}_n = \begin{bmatrix} a_{n,0} & a_{n,1} & \dots & a_{n,L_A-1} \end{bmatrix}^T, \tag{8}$$

and the NL_A-dimensional vector of stacked coefficient vectors is given by

$$\mathbf{a} = \begin{bmatrix} \mathbf{a}_1^T & \mathbf{a}_2^T & \dots & \mathbf{a}_N^T \end{bmatrix}^T. \tag{9}$$

Obviously, the signals played by the loudspeakers will also propagate to the eardrum, i.e.,

$$t[k] = \mathbf{D}^T(q)\mathbf{u}[k], \tag{10}$$

where $\mathbf{D}(q)$ is the N-dimensional vector of the acoustic transfer functions between the loudspeakers and the eardrum, i.e.,

$$\mathbf{D}(q) = \begin{bmatrix} D_1(q) & D_2(q) & \dots & D_N(q) \end{bmatrix}^T. \tag{11}$$

The L_D-dimensional coefficient vector of the impulse response of $D_n(q)$ for the n-th loudspeaker is defined as

$$\mathbf{d}_n = \begin{bmatrix} d_{n,0} & d_{n,1} & \dots & d_{n,L_D-1} \end{bmatrix}^T. \tag{12}$$

In the frequency-domain, the spatial pre-processor response for the acoustic feedback paths can be computed by applying the N_{FFT}-point discrete Fourier transform (DFT) to the spatial pre-processor response in the time-domain, i.e.,

$$\mathbf{V}^H(\omega_n)\mathbf{A}(\omega_m) = \mathfrak{f}^T(\omega_n)\bar{\mathbf{V}}\mathbf{a}, \tag{13}$$

where ω_m is the m-th discrete angular frequency and $\mathfrak{f}(\omega_m)$ is the $(L_V + L_A - 1)$-dimensional vector of the DFT matrix, i.e.,

$$\mathfrak{f}(\omega_m) = \begin{bmatrix} 1 & e^{-\frac{j2\pi m}{N_{FFT}}} & \dots & e^{-\frac{j2\pi m(L_V+L_A-2)}{N_{FFT}}} \end{bmatrix}^T, \tag{14}$$

and $\bar{\mathbf{V}}$ is the $(L_V + L_A - 1) \times NL_A$-dimensional matrix of the concatenated $(L_V + L_A - 1) \times L_A$-dimensional convolution matrices $\bar{\mathbf{V}}_n$, i.e.,

$$\bar{\mathbf{V}} = \begin{bmatrix} \bar{\mathbf{V}}_1 & \dots & \bar{\mathbf{V}}_N \end{bmatrix}. \tag{15}$$

Before designing the spatial pre-processor $\mathbf{A}(q)$ for the N loudspeakers in Section 4, we first analyze the transfer functions of the considered hearing device system in the next section.

3. SYSTEM ANALYSIS

By combining (1), (5), and (6) we can rewrite the microphone signal $y[k]$ as

$$y[k] = \frac{1}{1 - G(q)\mathbf{V}^T(q)\mathbf{A}(q)} x[k]. \tag{16}$$

Hence, perfect feedback suppression can be achieved when $y[k] = x[k]$, i.e., the spatial pre-processor cancels the loudspeaker contributions at the microphone position, i.e.,

$$\mathbf{V}^T(q)\mathbf{A}(q) = 0. \tag{17}$$

When this condition is satisfied, the resulting signal at the eardrum is equal to

$$t[k] = \mathbf{D}^T(q)\mathbf{A}(q)G(q)x[k]. \tag{18}$$

Thus, the spatial pre-processor $\mathbf{A}(q)$ obviously modifies the desired signal at the eardrum, possibly leading to perceivable distortions.

In the remainder of this paper, we assume that J different measurements of the acoustic feedback paths $\mathbf{V}_j(q)$, $j = 1,\dots,J$, are available and that the hearing device forward path is a broadband gain, i.e., $G(q) = |G|q^{-d_G}$, with d_G a delay. The maximum stable gain (MSG) $\mathcal{M}_j$ for the j-th set of acoustic feedback path measurements is given by

$$\mathcal{M}_j = \frac{1}{\max_{\omega_m} |\mathbf{V}_j^H(\omega_m)\mathbf{A}(\omega_m)|^2}. \tag{19}$$

Note that here we assume that the phase of the closed-loop system transfer function is a multiple of 2π for all frequencies, i.e., the worst-case assumption for the MSG. Assuming that for multiple sets of acoustic feedback paths, the worst maximum stable gain determines the MSG of the hearing device, we define the overall MSG as

$$\mathcal{M} = \min_{j} \mathcal{M}_j. \tag{20}$$

4. ACTIVE FEEDBACK SUPPRESSION

In this section we propose to design the spatial pre-processor $\mathbf{A}(q)$ maximizing the overall MSG in (20) while limiting the distortions of the signal arriving at the eardrum in (18). This spatial pre-processor is able to reduce acoustic feedback by producing destructive interference of the multiple loudspeaker signals at the position of the microphone, hence yielding similarities to active noise control approaches [17, 18]. It should be noted, however, that since we do not aim at reducing (unknown) sounds received by microphone from the outside but at reducing (known) playback signals, the typical low-latency requirements associated with active control algorithms do not apply here.

In the following we assume that both the acoustic feedback paths $\mathbf{V}_j(q)$ as well as multiple acoustic transfer functions to the eardrum $\mathbf{D}_i(q)$, $i = 1,...,I$ are known, e.g., by measurement. Note that while the acoustic feedback paths can be easily measured, the acoustic transfer functions to the eardrum need to be estimated in practice, e.g., using an electro-acoustic model [23] or using a practically feasible microphone placement at the inside of the hearing device.

Using (19) and (20), the coefficient vector $\mathbf{a}$ of the spatial pre-processor that maximizes the overall MSG can be obtained by minimizing the following cost function

$$\boxed{J_{MM}(\mathbf{a}) = \max_{\omega_m, j} |\mathbf{V}_j^H(\omega_m)\mathbf{A}(\omega_m)|^2} \tag{21}$$

where $\mathbf{V}_j^H(\omega_m)\mathbf{A}(\omega_m)$ is defined similarly as $\mathbf{V}^H(\omega_m)\mathbf{A}(\omega_m)$ in (15) for the jth set of acoustic feedback path measurements. In order to avoid the trivial solution $\mathbf{a} = \mathbf{0}$, we add a constraint aiming at a distortionless desired signal $D_{n_0,i}(q)\tilde{u}[k]$ at the eardrum. Using (18) this can be formulated as

$$\bar{\mathbf{D}}_i \mathbf{a} = \bar{\mathbf{d}}_{n_0,i}. \tag{22}$$

where $\bar{\mathbf{D}}_i$ is the $(L_D + L_A - 1) \times NL_A$-dimensional matrix of concatenated $(L_D + L_A - 1) \times L_A$-dimensional convolution matrices $\bar{\mathbf{D}}_{i,n}$, $n = 1,...,N$, i.e.,

$$\bar{\mathbf{D}}_i = \begin{bmatrix} \bar{\mathbf{D}}_{1,i} & \dots & \bar{\mathbf{D}}_{N,i} \end{bmatrix}, \tag{23}$$

and $\bar{\mathbf{D}}_{n,i}$ is the $(L_D + L_A - 1) \times L_A$-dimensional convolution matrix of $\mathbf{d}_{n,i}$, and $\bar{\mathbf{d}}_{n,i}$ is the $(L_D + L_A - 1)$-dimensional zero-padded vector of $\mathbf{d}_{n,i}$. Assuming the availability of I sets of acoustic path measurements between the loudspeakers and the eardrum, the constraints in (22) can be reformulated as

$$\boxed{J_{ls}(\mathbf{a}) = \sum_{i=1}^{I} \|\bar{\mathbf{D}}_i \mathbf{a} - \mathbf{d}_{n_0,i}\|_2^2 = 0} \tag{24}$$

where $J_{ls}(\mathbf{a})$ denotes the distortion cost function. Since in practice some distortions of the signal arriving at the eardrum can be tolerated, we propose to minimize the regularized cost function

$$\boxed{J_{MM}^{reg}(\mathbf{a}) = J_{MM}(\mathbf{a}) + \bar{\lambda} J_{ls}(\mathbf{a})} \tag{25}$$

where $\bar{\lambda} = \lambda \frac{\text{trace}\{\sum_{j=1}^{J} \bar{\mathbf{V}}_j^T \bar{\mathbf{V}}_j\}}{\text{trace}\{\sum_{i=1}^{I} \{\bar{\mathbf{D}}_i^T \bar{\mathbf{D}}_i\}}$ is a real-valued trade-off parameter, that allows to trade of between feedback suppression and distortionless transmission of the loudspeaker signal. More in particular, by the definition of $J_{ls}(\mathbf{a})$ the trade-off parameter allows to trade-off between the following cases:

1. $\lambda \to \infty$: perfect preservation of the desired signal at the eardrum by using only the reference loudspeaker n_0 and limited (or no) feedback suppression.
2. $\lambda \to 0$: perfect feedback suppression by the trivial solution $\mathbf{a} = \mathbf{0}$ but no loudspeaker playback, which is obviously not desired.
3. $0 < \lambda < \infty$: increasing feedback suppression at the cost of increasing distortion of the desired signal at the eardrum.

The optimization problem in (25) can be approximated by using the real rotation theorem [24]. Briefly, the real rotation theorem states that the absolute value of a complex scalar can be approximated with arbitrarily small error by projecting the complex value onto a rotating complex pointer using a finite set of rotation angles. Using the real-rotation theorem, the optimization problem in (25) can then be approximated as the following quadratically constrained quadratic programming (QCQP) problem

$$\min_{t,\xi,\mathbf{a}} \quad t + \xi \tag{26a}$$

$$\text{subject to} \quad p_j(\omega_m)\cos\phi_l + q_j(\omega_m)\sin\phi_l \leq t, \quad \forall \omega_m, j, l \tag{26b}$$

$$\bar{\lambda} \sum_{i=1}^{I} \|\bar{\mathbf{D}}_i \mathbf{a} - \mathbf{d}_{n_0,i}\|_2^2 \leq \xi, \tag{26c}$$

where t and ξ are auxiliary variables, ϕ_l is the lth rotation angle, $l = 1,...,N_\phi$, and $p_j(\omega_m)$ and $q_j(\omega_m)$ are the real and the imaginary parts of the residual spatial pre-processing error, i.e.,

$$p_j(\omega_m) = \mathfrak{Re}\{\mathbf{V}_j^H(\omega_m)\mathbf{A}(\omega_m)\}, \tag{27}$$

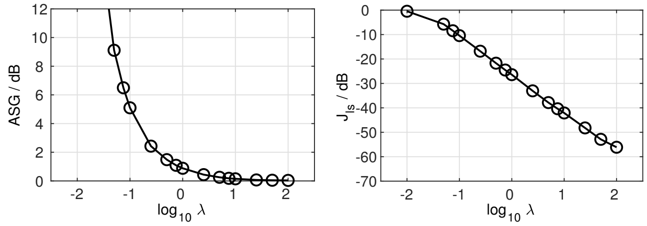
$$q_j(\omega_m) = \mathfrak{Im}\{\mathbf{V}_j^H(\omega_m)\mathbf{A}(\omega_m)\}. \tag{28}$$

The QCQP problem in (26) can be efficiently solved using existing convex optimization tools, e.g., CVX [21, 22].

5. EXPERIMENTAL EVALUATION

In this section the performance of the proposed active feedback suppression algorithm using multiple loudspeakers is investigated. Specifically, we evaluate the ability to suppress the acoustic feedback at the microphone position as well as the preservation of the desired signal at the eardrum. To this end, we also consider the use of a microphone in the ear canal to estimate the acoustic transfer functions to the eardrum.

Acoustic feedback paths and acoustic transfer functions were measured using a custom earpiece with two loudspeakers and four microphones [19] on a GRAS 45BB-12 KEMAR Head & Torso with low-noise ear simulators in an anechoic room. In this evaluation we only use a single microphone, more in particular, the entrance microphone located in close vicinity to the outer vent opening. The impulse responses were sampled at 16 kHz and truncated to length $L_V = 100$ and $L_D = 100$ for the acoustic feedback paths and the acoustic transfer functions, respectively. All acoustic impulse responses were measured five times after refitting of the earpiece with a telephone receiver in close distance to the ear and in free-field, i.e., without any obstruction close to the dummy head, resulting in ten different sets of acoustic feedback path measurements. In all experiments we used $N = 2$ loudspeakers and $n_0 = 1$, i.e., the inner loudspeaker in Figure 1. For the design of the spatial pre-processor $\mathbf{A}(q)$ we used a filter length of $L_A = 40$, an FFT size of $N_{FFT} = 512$, and $N_\phi = 16$ discrete rotation angles for the QCQP in (26). We used



(a) ASG. (b) Distortion.

Figure 3: Added stable gain and distortion for the optimal performance in Experiment 1.

a forward path $G(q) = G_0 q^{-d_G}$, with $d_G = 96$ a delay and G_0 a broadband gain such that the hearing device using only the reference loudspeaker was approximately overcritical, i.e., $|G(q)V_{n_0}(q)| \approx 1$.

The feedback suppression performance is evaluated using the added stable gain, which is computed as

$$ASG = 10\log_{10} \frac{\max\limits_{\omega_m} |V_{n_0}(\omega_m)|^2}{\max\limits_{\omega_m} |\mathbf{V}^H(\omega_m)\mathbf{A}(\omega_m)|^2}. \tag{29}$$

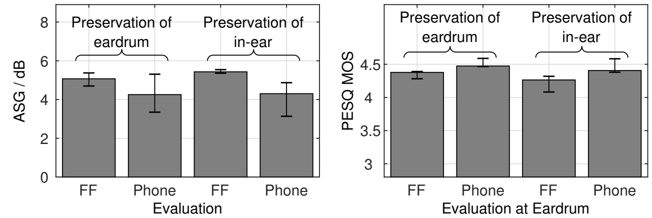
Note that as in (19), we make the worst-case assumption that the phase of the closed-loop transfer function is a multiple of 2π. The preservation of the loudspeaker signal at the eardrum is evaluated using the perceptual evaluation of speech quality measure (PESQ) [25]. For the PESQ measure we used a 80 s long speech signal comprising different male and female speakers. As reference signal we used $D_{n_0}(q)G(q)x[k]$ and as a test signal we used $t[k]$. Note that similarly as in [16], to avoid the feedback components to influencing the quality evaluation, we assumed the feedback paths to be absent.

5.1. Experiment 1: Influence of trade-off parameter

In the first experiment, we evaluate the influence of the trade-off parameter λ in (25) on the feedback suppression performance and preservation of the loudspeaker signal. We compute a single spatial pre-processor using $J = 5$ sets of acoustic feedback paths and $I = 5$ sets of acoustic transfer functions, all measured in free-field. Figure 3 shows the median ASG across all five sets of acoustic feedback paths and the values of J_{ls} defined in (24) as a function of the trade-off parameter. As can be observed, decreasing λ yields an increase in ASG due to the usage of the second loudspeaker at the cost of an increase in distortion. Furthermore, for large values of λ, the ASG stabilizes around 0 dB while the distortion further decreases. This is due to spatial pre-processor effectively only selecting the reference loudspeaker. Both observations are in line with the theoretical observations in Section 4. Based on the results in Figure 3, in the following experiment we will use a trade-off parameter $\lambda = 10^{-1}$, limiting the distortion of the desired signal at the eardrum to approximately -10 dB.

5.2. Experiment 2: Robustness

In the second experiment, we evaluate the sensitivity of the spatial pre-process to variations in both the acoustic feedback paths as well as to the estimation of the transfer function to the eardrum using an in-ear microphone placement in the earpiece. Therefore, we use $J = 4$ feedback paths measurements in free-field and their corresponding $I = 4$ measurements of the (estimated) acoustic transfer function to the eardrum. For each of the five sets of measurements a different spatial pre-processor is computed using the remaining $J = I = 4$ sets of measurements. To



(a) ASG. (b) PESQ.

Figure 4: Evaluation results of Experiment 2 in terms of (a) the median ASG and (b) the median PESQ MOSs. Errorbars show minimum and maximum values, respectively. Results are shown for both conditions when the acoustic path between the loudspeakers and the eardrum is known (preservation of eardrum) and when the acoustic path between the loudspeaker and the eardrum is estimated using an in-ear microphone (preservation of in-ear).

evaluate the performance, we compute the performance measures for each of the five spatial pre-processors using the free-field and the corresponding phone measurements, i.e., using a leave-one-out cross validation approach.

Figure 4 shows the results in terms of the median ASG and the median PESQ mean opinion scores (MOSs). When the acoustic transfer function to the eardrum is known can be preserved, a robust ASG of approximately 5 dB in free-field and 4–5 dB in the presence of a telephone receiver can be achieved (cf. Figure 4a). At the same time the quality of the signal at the eardrum is of very high quality with PESQ MOSs larger than 4.3 (cf. Figure 4b). When the acoustic transfer function to the eardrum is unknown and estimated by measuring the acoustic transfer functions to the in-ear microphone of the hearing device (cf. Figure 1), a very similar performance in terms of the ASG is obtained (cf. Figure 4a) while the median PESQ MOS scores are still larger than 4.2.

In conclusion, these results show that by using a second loudspeaker in a hearing device, a robust ASG can be achieved without significantly distorting desired signal at the eardrum. This is the case even when the acoustic transfer function to the eardrum is estimated using an in-ear microphone.

6. CONCLUSION

In this paper, we proposed to suppress acoustic feedback in a hearing device by exploiting the availability of multiple loudspeakers. The multi-loudspeaker feedback suppression filters were computed by maximizing the maximum stable gain of the hearing device, while limiting the distortions of the desired signal at the eardrum. Using the real rotation theorem, the resulting optimization problem was formulated as a quadratically constrained quadratic programming problem. Experimental results using measured acoustic feedback paths from a custom hearing device showed that by using two loudspeakers the feedback contribution in the microphone can be robustly reduced. We showed that by using an additional loudspeaker in a hearing device the maximum stable gain can be increased by approximately 4–5 dB without significantly distorting the desired signal at the eardrum, even when the acoustic transfer function to the eardrum is not exactly known.

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