Multi-loudspeaker equalization for acoustic transparency in a custom hearing device

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Abstract

To improve the sound quality of hearing devices, equalization algorithms can be used that aim at achieving acoustic transparency, i.e., listening with the device in the ear is perceptually similar to the open ear. The equalization filter needs to ensure that the superposition of the processed and equalized signal played by the device and the signal leaking through the device into the ear canal matches a processed version of the signal reaching the eardrum of the open ear. Since equalization using a single loudspeaker typically does not allow for perfect equalization, in this paper we propose to use a multi-loudspeaker equalization filter to achieve acoustic transparency in a custom multi-loudspeaker hearing device. The equalization filter is computed by minimizing a regularized least-squares optimization problem. Experimental results using measured acoustic transfer functions show that the proposed multi-loudspeaker equalization filter is able to provide the desired signal at the eardrum for different gains and delays of the hearing device.

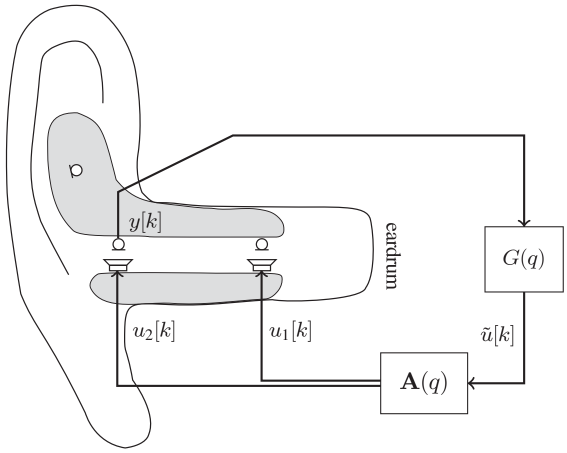

Figure 1: Schematic overview of considered single-microphone multi-loudspeaker custom hearing earpiece setup.

1 Introduction

Despite major improvements in hearing device technology in the past decades, the acceptance of hearing aids and assistive listening devices is still rather limited, partly due to poor sound quality [1, 2]. This is most prominent in first-time users and users with normal hearing or mild-to-moderate hearing loss. While these users would benefit from advanced algorithms like noise reduction, dereverberation and dynamic range compression, they usually do not accept a degradations of the sound quality. In order to improve the sound quality, equalization algorithms have been proposed that aim at achieving so-called acoustic transparency [3, 4, 6, 7], i.e., listening with the device in the ear achieves a similar perceptual impression as listening without the device.

Generally, equalization algorithms for acoustic transparency aim at matching the sound pressure reaching the eardrum when the ear is occluded by the device with the sound pressure at the eardrum of the open ear [7]. When the device is inserted in the ear, the sound pressure at the eardrum consists of the superposition of the direct sound leaking into the (partly) occluded ear canal and the sound picked up by the microphone of the device, processed and played by the device. For the open ear, the sound pressure at the eardrum only consists of the direct sound. Equalization in hearing devices is commonly performed using a single loudspeaker [6], i.e., a single equalization filter is computed to match the sound pressure of the occluded ear and the open ear.

Computing the equalization filter usually requires the inversion of the acoustic transfer function (ATF) of the signal picked up by the microphone of the device. However, since this ATF typically has zeros inside and outside the unit circle, perfect inversion (with a stable and causal filter) cannot be achieved [8, 9]. Hence, approximate solutions are required to obtain a good equalization filter when using a single loudspeaker [3–7]. On the contrary, using multiple loudspeakers perfect equalization can be achieved when the conditions of the multiple-input/output inverse theorem (MINT) are satisfied [9]. Briefly, MINT states that perfect inversion of a multi-channel system can be achieved if all channels are co-prime, i.e., they do not share common zeros, and the equalization filters are of sufficient length.

Therefore, in this paper we consider equalization using multiple loudspeakers to achieve acoustic transparency, which has similarities with approaches for listening room compensation, e.g., [10, 11]. Specifically, we consider a custom earpiece with three microphones and two loudspeakers [7, 12], depicted in Figure 1. We design the multi-loudspeaker equalization filter by minimizing a least-squares cost function and show that for the considered scenario the multi-loudspeaker system exhibits common zeros and hence MINT is not directly applicable. Since the common zeros of the system are exactly known, we use this knowledge and reformulate the optimization problem accordingly. Finally, since the ATFs between the loudspeakers and the eardrum are likely to exhibit near-common zeros we use regularization [13, 14] to obtain the equalization filters.

Experimental results using individually measured acoustic transfer functions from the custom multi-loudspeaker earpiece in Figure 1 show that the proposed multi-loudspeaker equalization approach is able to achieve almost perfect equalization. Furthermore, we show that the equalization performance depends on the gain and the processing delay of the hearing device.

2 Scenario and Problem Statement

Consider a single-microphone multi-loudspeaker hearing device with N loudspeakers as depicted in Figure 2. For simplicity we assume that all transfer functions are linear and time-invariant and that they can be modeled as polynomials in the variable q [15]. We assume that the signal $y[k]$ picked up by the microphone of the device is the signal emitted from a single directional sound source $s[k]$, i.e.,

$$y[k] = H_m(q)s[k], \tag{1}$$

where k denotes the discrete time index and $H_m(q)$ is the ATF between the sound source and the microphone of the hearing device $H_m(q) = \mathbf{h}_m^T\mathbf{q}$. The L_H-dimensional impulse response (IR) vector of $H_m(q)$ is given by

$$\mathbf{h}_m = [h_{m,0} \quad h_{m,1} \quad \dots \quad h_{m,L_H-1}]^T, \tag{2}$$

and the $\mathbf{q}$ is the vector of delay elements q of appropriate length. The microphone signal is processed by the forward path $G(q)$ of

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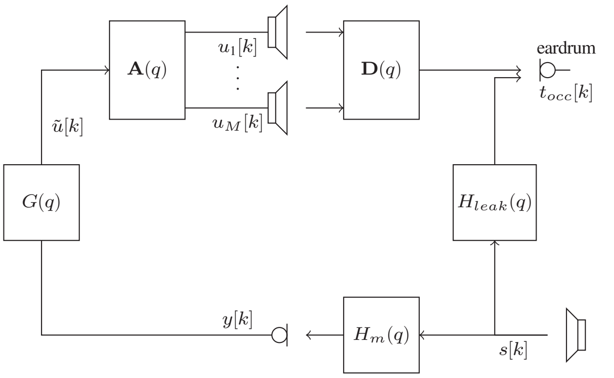

Figure 2: Generic single-microphone multi-loudspeaker hearing device setup considered in this work.

the hearing device, with L_G-dimensional IR vector $\mathbf{g}$, yielding the intermediate signal $\tilde{u}[k]$, i.e.,

$$\tilde{u}[k] = G(q)y[k]. \tag{3}$$

This intermediate signal is then used as an input to N different equalization filters $A_n(q)$, $n = 1,...,N$, yielding the N-dimensional loudspeaker signal vector $\mathbf{u}[k]$, i.e.,

$$\mathbf{u}[k] = \begin{bmatrix} A_1(q) \\ \vdots \\ A_N(q) \end{bmatrix} \tilde{u}[k] = \mathbf{A}(q)\tilde{u}[k]. \tag{4}$$

The L_A-dimensional equalization filter coefficient vector of $A_n(q)$ is defined as

$$\mathbf{a}_n = [a_{n,0} \quad a_{n,1} \quad ... \quad a_{n,L_A-1}]^T. \tag{5}$$

Furthermore, we define the NL_A-dimensional vector of concatenated equalization filter coefficient vectors as

$$\mathbf{a} = \left[\mathbf{a}_1^T \quad ... \quad \mathbf{a}_N^T\right]^T. \tag{6}$$

For the occluded ear, i.e., when the device is inserted and processing the microphone signal, the signal $t_{occ}[k]$ reaching the eardrum of the listeners is the superposition of the multiple loudspeaker signals and the signal leaking into the (partially) occluded ear canal, i.e.,

$$t_{occ}[k] = \mathbf{D}^T(q)\mathbf{u}[k] + H_{leak}(q)s[k], \tag{7}$$

where $H_{leak}(q)$ denotes the ATF between the source and the eardrum for the occluded ear, with L_H-dimensional IR vector $\mathbf{h}_{leak}$. The N-dimensional vector $\mathbf{D}(q)$ contains the ATFs between the loudspeakers of the hearing device and the eardrum, i.e.,

$$\mathbf{D}(q) = [D_1(q) \quad ... \quad D_N(q)]^T, \tag{8}$$

with the L_D-dimensional IR vector of $D_n(q)$ denoted by $\mathbf{d}_n$.

For the open ear, the desired signal reaching the eardrum $t_{des}[k]$ is a processed version of the signal reaching the eardrum of the listeners, i.e.,

$$t_{des}[k] = G(q)\underbrace{H_{open}(q)s[k]}_{t_{open}[k]}, \tag{9}$$

where $H_{open}(q)$ denotes the ATF between the source and the open ear with L_H-dimensional IR vector $\mathbf{h}_{open}$. In order to achieve acoustic transparency, the goal is to obtain the equalization filter $\mathbf{A}(q)$ such that the signal $t_{occ}[k]$ in (7) is perceptually not distinguishable from the signal $t_{des}[k]$ in (9).

3 Transfer Function Analysis

In this section, we analyze the considered single-microphone multi-loudspeaker system in terms of its system transfer function between the source and the eardrum. For the occluded ear, the system transfer function is obtained by combining and rearranging (1), (3), (4), and (7), leading to

$$O_{occ}(q) = \frac{t_{occ}[k]}{s[k]} = \mathbf{D}^T(q)\mathbf{A}(q)G(q)H_m(q) + H_{leak}(q). \tag{10}$$

Similarly, for the open ear, the system transfer function is obtained from (9) as

$$O_{des}(q) = \frac{t_{des}[k]}{s[k]} = G(q)H_{open}(q). \tag{11}$$

By equating (10) and (11) we observe that the optimal equalization filter needs to fulfill

$$G(q)H_m(q)\mathbf{D}^T(q)\mathbf{A}(q) = G(q)H_{open}(q) - H_{leak}(q) \tag{12}$$

After rearranging, we observe that the optimal equalization filter similarly needs to fulfill

$$\mathbf{D}^T(q)\mathbf{A}(q) = \frac{H_{open}(q)}{H_m(q)} - \frac{H_{leak}(q)}{H_m(q)}\frac{1}{G(q)}. \tag{13}$$

Note that while the optimal filter in (12) depends on the ATFs the optimal filter in (13) depends on relative transfer functions (RTFs). Furthermore, the optimal filter depends on the hearing aid forward path $G(q)$. In order to assess this dependency, in the following we analyse two extreme cases observed from (13):

1. Assuming the absence of the leakage component ($H_{leak}(q) = 0$), the optimal equalizing filter aims at matching the first term on the right-hand-side of (13), i.e.,

$$\mathbf{D}^T(q)\mathbf{A}(q) = \frac{H_{open}(q)}{H_m(q)}, \tag{14}$$

   such that $O_{occ}(q) = G(q)H_{open}(q)$.

2. Assuming that $H_{open}(q) = 0$, the optimal equalization filter aims at actively suppressing the leaking component at the eardrum, i.e.,

$$\mathbf{D}^T(q)\mathbf{A}(q) = -\frac{H_{leak}(q)}{H_m(q)}\frac{1}{G(q)}, \tag{15}$$

   such that $O_{occ}(q) = 0$.

From the above analysis we conclude that for large forward path gains, equalization becomes more important than suppression of the leakage component, while for small forward path gains the leakage component dominates and needs to be suppressed. Furthermore, depending on the delay of $G(q)$, the desired transfer function after equalization can become increasingly acausal, which may impact the equalization performance.

4 Equalization filter design

In this section, we present a regularized least-squares based procedure to compute the multi-loudspeaker equalization filter $\mathbf{A}(q)$. We assume knowledge of all required ATFs, e.g., by measurement. Note that in practice an estimate of the ATF between the source and the eardrum for the open ear $H_{open}(q)$ can be obtained by an appropriate transformation of the ATF between the source and the microphone $H_m(q)$ [17]. In addition, an estimate of the ATFs between the loudspeakers of the hearing device and the eardrum $\mathbf{D}(q)$ can be obtained using an in-ear microphone and an electro-acoustic model [18]. We start by formulating the least-squares cost function optimizing the equalization filters according to (12) and show that the transfer functions to be equalized share exact common zeros. Since these common zeros are known, we use this knowledge and reformulate the optimization problem accordingly. Finally, we motivate the necessity to include additional regularization to obtain the equalization filters.

The equation of the optimal equalization filter in (12) can be reformulated using matrix and vector notation as

$$\mathbf{C}\mathbf{a}=\mathbf{v}, \tag{16}$$

where $\mathbf{C}$ is the $L_C+L_A-1 \times NL_A$-dimensional matrix, with $L_C=L_G+L_H+L_D-2$, defined as

$$\mathbf{C}=\mathbf{G}\mathbf{H}_m\mathbf{D}, \tag{17}$$

where $\mathbf{G}$ is the $(L_C+L_A-1)\times(L_H+L_D+L_A-2)$-dimensional convolution matrix of the IR vector $\mathbf{g}$, $\mathbf{H}_m$ is the $(L_H+L_D+L_A-2)\times(L_D+L_A-1)$-dimensional convolution matrix of the IR vector $\mathbf{h}_m$, $\mathbf{D}$ 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 $\mathbf{D}_n$ of the IR vector $\mathbf{d}_n$, i.e.,

$$\mathbf{D}=[\mathbf{D}_1 \quad \ldots \quad \mathbf{D}_N]. \tag{18}$$

Furthermore, $\mathbf{v}$ is the (L_C+L_A-1)-dimensional vector of the desired IR

$$\mathbf{v}=\mathbf{G}\tilde{\mathbf{h}}_{open}-\tilde{\mathbf{h}}_{leak}, \tag{19}$$

where $\tilde{\mathbf{h}}_{open}$ is the $(L_H+L_D+L_A-2)$-dimensional zero padded vector of the open ear IR vector $\mathbf{h}_{open}$ and $\tilde{\mathbf{h}}_{leak}$ is the (L_C+L_A-1)-dimensional zero-padded coefficient vector of the IR vector $\mathbf{h}_{leak}$.

The NL_A-dimensional equalization filter coefficient vector $\mathbf{a}$ is then obtained by minimizing the following least-squares cost function

$$J_{LS}(\mathbf{a})=\|\mathbf{C}\mathbf{a}-\mathbf{v}\|_2^2. \tag{20}$$

Assuming that $L_A \geq \frac{L_C-1}{N-1}$ and the matrix $\mathbf{C}$ in (20) is of full row-rank, the optimal solution minimizing (20) is obtained as

$$\mathbf{a}=\mathbf{C}^T(\mathbf{C}\mathbf{C}^T)^{-1}\mathbf{v}. \tag{21}$$

However, since the rows in $\mathbf{C}$ are linearly related by the matrices $\mathbf{G}$ and $\mathbf{H}_m$, the matrix $\mathbf{C}$ is not of full row-rank, such that the optimal solution in (21) can not be computed and thus perfect equalization can not be achieved. In order to mitigate this rank deficiency[1], we propose to left multiply both $\mathbf{C}$ and $\mathbf{v}$ by the pseudoinverse of $\mathbf{G}\mathbf{H}_m$, i.e.,

$$\tilde{\mathbf{C}}=(\mathbf{H}_m^T\mathbf{G}^T\mathbf{G}\mathbf{H}_m)^{-1}\mathbf{H}_m^T\mathbf{G}^T\mathbf{C}=\mathbf{D}, \tag{22}$$

$$\tilde{\mathbf{v}}=(\mathbf{H}_m^T\mathbf{G}^T\mathbf{G}\mathbf{H}_m)^{-1}\mathbf{H}_m^T\mathbf{G}^T\mathbf{v}, \tag{23}$$

which is similar to writing (12) using matrix and vector notation. Note that $\tilde{\mathbf{v}}$ represents a relative transfer function, which cannot be perfectly modeled using an FIR filter and hence perfect equalization is not possible. Nevertheless, assuming that the matrix $\mathbf{D}$ (and hence $\tilde{\mathbf{C}}$) is of full row-rank, using $\tilde{\mathbf{C}}$ and $\tilde{\mathbf{v}}$ instead of $\mathbf{C}$ and $\mathbf{v}$ in (20) solves the problem of common zeros and choosing $L_A \geq \frac{L_D-1}{N-1}$ allows to apply MINT. However, the ATFs $\mathbf{D}(q)$ between the loudspeaker of the hearing device and the eardrum are likely to share near-common zeros due the close proximity of the loudspeakers in the considered hearing device (cf. Figure 1). This may lead to ill-conditioning of the matrix inversion when using $\tilde{\mathbf{C}}$ in (21). In order to mitigate this ill-conditioning, we additionally use regularization [13, 14] to obtain the equalization filter, i.e., we reformulate the least-squares optimization problem in (20) using $\tilde{\mathbf{C}}$ and $\tilde{\mathbf{v}}$ as

$$\boxed{J_{rLS}(\mathbf{a})=\|\tilde{\mathbf{C}}\mathbf{a}-\tilde{\mathbf{v}}\|_2^2+\lambda\|\mathbf{a}\|_2^2} \tag{24}$$

where λ is a real-valued non-negative trade-off parameter. The optimal solution minimizing $J_{rLS}(\mathbf{a})$ is equal to

$$\boxed{\mathbf{a}_{rLS}=(\tilde{\mathbf{C}}^T\tilde{\mathbf{C}}+\lambda\mathbf{I})^{-1}\tilde{\mathbf{C}}^T\tilde{\mathbf{v}}} \tag{25}$$

where $\mathbf{I}$ is the identity-matrix of appropriate dimensions. Here we choose $\lambda=10^{-8}$ to guarantee numerically stable inversion of $\tilde{\mathbf{C}}^T\tilde{\mathbf{C}}$.

[1]Note that regularization could also be used to mitigate the rank-deficiency here. However, we have perfect knowledge (in terms of the convolution matrices) of the common zeros and hence choose to exploit this knowledge.

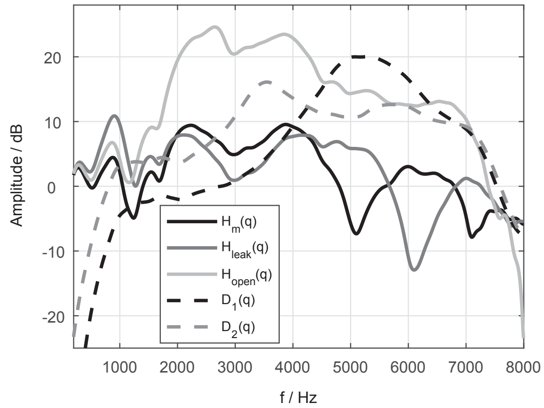

Figure 3: Amplitude responses of the acoustic transfer functions used in the experimental evaluation.

5 Experimental Evaluation

In this section we evaluate the proposed multi-loudspeaker equalization approach using measured ATFs from custom earpiece [7] in Figure 1 with $N=2$ loudspeakers. Specifically, we evaluate the impact of the delay and gain of the forward path on the equalization performance. Note that although the earpiece has three microphones, here we only consider the microphone located at the outer face of the venting tube, providing a natural position for sound pickup. All ATFs were measured in an anechoic chamber on a human subject as described in [12, 16], resampled to 16 kHz and truncated to length $L_H=200$, $L_D=100$. The directional sound source was positioned approximately 1 m in front of the subject and measurements of the pressure at the eardrum were obtained using a probe tube microphone. Figure 3 shows the amplitude response of the different ATFs used in the experimental evaluation. For the forward path we used a broadband gain G_0 and a delay of $d_G=L_G-1$ samples, i.e., $G(q)=10^{G_0/20}q^{-d_G}$. In all experiments we use an equalization filter length of $L_A=100$.

In order to evaluate the performance of the proposed multi-loudspeaker equalization approach, we will consider the amplitude responses of the system transfer functions for the processed open ear $O_{des}(q)$ in (11), and the system transfer function for the (equalized) occluded ear $O_{occ}(q)$ in (10). Furthermore, in order to assess the perceptual quality of the signal at the eardrum we use a speech intelligibility weighted signal distortion (SD_{int}) measure and the perceptual evaluation of speech quality (PESQ) measure [20]. The speech intelligibility weighted SD_{int} measure is computed similar as in [21] as

$$SD_{int}=\sqrt{\int_{f_l}^{f_u}\frac{w_{int}(f)}{\Delta f}\left(20\log_{10}\left|\frac{O_{occ}(f)}{O_{des}(f)}\right|\right)^2 \mathrm{d}f}, \tag{26}$$

where f is the frequency, $f_l=0$ Hz and $f_u=8000$ Hz are the lower and upper frequency bounds, w_{int} and Δf are the one-third-octave weighting and bandwidth according to [22]. For the PESQ measure we use the equalized signal $t_{occ}[k]$ at the eardrum as the test signal and the processed signal of the open ear canal $t_{des}[k]$ in (9) as the reference signal. As speech signal we use an 80 s long signal of different speakers from the TIMIT database [19] that has previously been used in [23].

In the following, we inspect the amplitude responses of the processed open-ear system transfer function $O_{des}(q)$ and the (equalized) occluded ear system transfer function $O_{occ}(q)$. First, we investigate the impact of different broadband gains G_0 in the forward path for a fixed forward path delay d_G and second, we investigate the impact of different forward path delays d_G for a fixed forward path gain G_0.

Figure 4 shows the amplitude responses of the processed open ear system transfer function and the equalized occluded transfer function for different forward path gains G_0 using a forward path delay $d_G=2$. Generally, the equalized occluded ear system transfer function matches the desired processed open ear transfer function very well for all considered gains. Comparing $N=1$ (computed using (25)) and $N=2$ shows that using the proposed multiple loudspeaker equalization results in better equalization of the sound pressure. While when using $N=2$ loudspeakers for the negative gain of $G_0=-10$ dB

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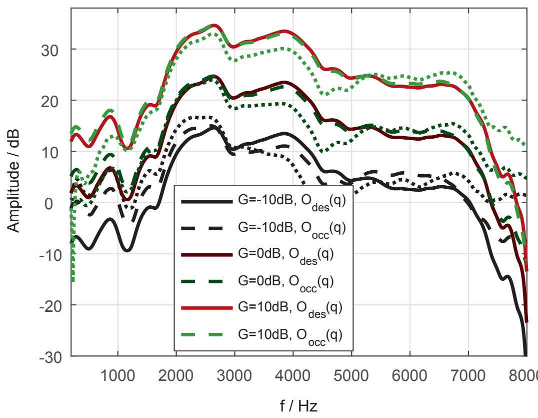

Figure 4: Amplitude responses of the system transfer functions for the occluded ear $O_{occ}(q)$ and the processed open ear $O_{des}(q)$ for different gains G_0 of the forward path using a forward path delay $d_G = 2$ and $N = 2$ loudspeakers. Dotted lines indicate the system transfer functions for the occluded ear $O_{occ}(q)$ when using $N = 1$ loudspeaker.

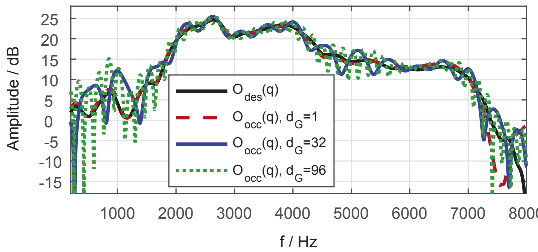

Figure 5: Amplitude responses of the system transfer functions for the occluded ear $O_{occ}(q)$ and the processed open ear $O_{des}(q)$ for different delays d_G of the forward path using a forward path gain $G_0 = 0$ dB.

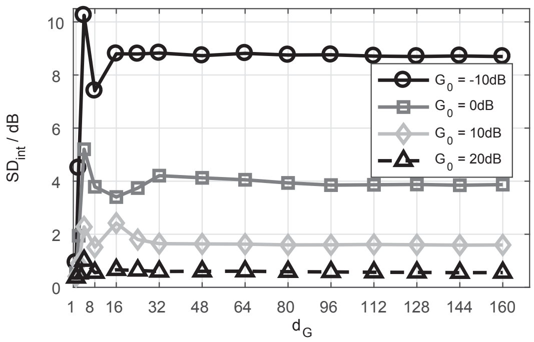

Figure 6: Speech intelligibility weighted signal distortion as a function of the forward path delay d_G for different forward path gains G_0.

the largest deviation from the desired processed open ear system transfer function is observed, for the largest gain of $G_0 = 10$ dB the best match is observed. This indicates that the proposed equalization works best if the forward path gain is large enough such that the output of the hearing device is large compared to the leakage sound component, i.e., in this case for $G_0 \geq 0$ dB (cf. Section 3).

Figure 5 shows the amplitude responses of the processed open-ear system transfer function and the equalized occluded ear system transfer function for different forward path delays L_G using a fixed forward path gain $G_0 = 0$ dB. The general shape of the desired transfer function is matched quite well, independent of the forward path delay. However, with increasing forward path delay comb-filtering effects are introduced that are visible as modulations of the amplitude response. While these are rather small for a delay of $d_G = 32$, they are more distinct for a more practical delay of $d_G = 96$ (corresponding to 6 ms) and may impact the perceived quality as investigated in the next paragraph. This is in line with the

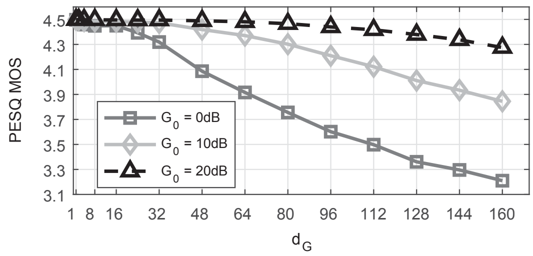

Figure 7: PESQ MOS values of the occluded ear signal at the ear drum $t_{occ}[k]$ as a function of the forward path delay d_G for different forward path gains G_0.

analysis in Section 3, where we concluded that for lower gains the equalization filter aims at actively suppressing the leakage component. As expected, with increasing forward path delay, the desired transfer function of the equalized system exhibits a larger acausal delay as can be seen from (15), where the equalized transfer function depends on the inverse of the forward path transfer function. Thus, the equalization performance is generally expected to be reduced.

To investigate the impact of both the forward path delay and the forward path gain on the perceived quality of the equalized signal at the eardrum, Figures 6 and 7 shows the SD_{int} results and PESQ mean opinion scores (MOS) for different forward path gains as a function of the forward path delay. While the SD_{int} measure in Figure 6 indicates larger distortion with decreasing forward path gain, for forward path delays $d_G \geq 48$ the performance stays rather constant. However, informal listening tests confirm an expected increased impact of comb-filtering effects with increasing delay (cf. Figure 5) which cannot be resolved by the SD_{int} measure. This decrease is reflected in evaluations using PESQ, where Figure 7 shows that for low forward path delays $d_G \leq 48$, the PESQ MOS values are larger than approximately 4.1, indicating a high perceptual speech quality of the equalized signal at the eardrum. Increasing the forward path delay generally leads to a reduction in the perceptual quality as indicated by the smaller PESQ MOS values. Furthermore, it can be observed from both SD_{int} and PESQ MOS that the best performance is obtained for the largest considered forward path gain $G_0 = 20$ dB, which is in line with the results presented in Figure 4. The main reason for the lower SD_{int} values and larger PESQ MOS values for larger forward path gains $G_0 \geq 0$ dB is the stronger sound pressure of the equalized sound compared to the leakage component (cf. Section 3). Hence, a better equalization performance is expected since the equalized sound component dominates the leakage sound component and consequently comb filtering effects are reduced and hence larger forward path delays can be applied.

In conclusion, the presented results show the applicability of the proposed multi-loudspeaker equalization approach to achieve acoustic transparency. Future research aims at validating the obtained results using formal subjective listening tests as well as improving the equalization performance to reduce comb filtering effects.

6 Conclusion

In this paper we proposed a multi-loudspeaker equalization approach to achieve acoustic transparency in a custom multi-loudspeaker earpiece. The equalization filter is computed based on a least-squares cost function. An analysis of the least-squares optimization problem revealed that perfect equalization is not possible since the transfer functions to be equalized share common zeros, resulting in the inversion of a rank-deficient matrix. Since these common zeros are known, we used this knowledge and reformulate the optimization problem accordingly. Since the acoustic transfer functions between the loudspeakers of the hearing device and the eardrum are likely to share near-common zeros, we incorporated regularization to compute the equalization filter. Experimental results using measured ATFs from a custom earpiece show that using the proposed multi-loudspeaker equalization filter design acoustic transparency can be achieved for a large range of forward path gains. While the equalization performance appears to be generally robust to forward path delays, comb filtering effects occur with practical forward path delays that should be addressed in future research.

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