Equalization filter design for achieving acoustic transparency in a semi-open fit hearing device

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Abstract

Acoustically transparent hearing devices should allow hearing equivalent to the open ear while providing the possibility to modify the sound reaching the eardrum in a desired manner. To this end, the output of the device is processed by means of an equalization filter, such that the superposition of the sound played back by the device and an acoustic sound component directly leaking into the ear canal approximates the transfer function to the open eardrum. A particular difficulty in designing the equalization filter is the occurrence of comb filtering effects due to a superposition of the direct sound and the delayed output of the device. Here, we propose a regularized least-squares design approach with a closed-form solution that takes into account individually measured transfer functions of the device and ear, as well as the processing delay. Experimental results utilizing measured transfer functions from a custom prototype device show good equalization performance, particularly a reduction of comb-filtering effects as a result of an automated frequency-dependent regularization.

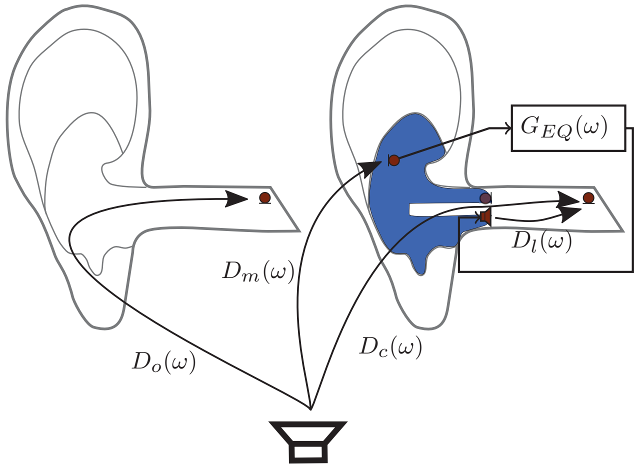

Figure 1: Considered acoustic scenario with acoustic transfer functions and signal processing blocks.

1 Introduction

Despite a great improvement in hearing technology in the past decades, the acceptance of hearing assistive devices is still limited, also due to a lack in sound quality [1, 2]. This is particularly true in potential first-time users with a mild-to-moderate hearing loss or even (near-to) normal hearing. Although they would benefit from hearing aid features like speech enhancement or amplification in acoustically challenging situations, they are usually not willing to accept a general degradation of the listening quality. To overcome this issue, several contributions aimed at constructing acoustically transparent devices [3–5]. Such devices allow a listening experience that is the same as with the open ear while having the possibility to modify the sound reaching the eardrum in a desired manner.

To achieve acoustic transparency, the transfer function between external sound sources and the eardrum should be the same for the open ear and the aided case, i.e., with the device inserted. To match the two cases, a so-called equalization filter needs to be computed that spectrally adjusts the output of the device. While in the unaided case, the transfer function includes only the direct sound component, in the aided case the transfer function is a superposition of the device's transfer function and a direct sound component that directly leaks into the partly occluded ear canal. The direct sound is present particularly in *semi-open* fit devices that include a vent to reduce the occlusion effect and improve the wearing comfort [6]. Since in digital devices the sound played back is usually delayed compared to the direct sound by some milliseconds, distortions occur due to comb filtering effects [7]. However, in previous approaches the contribution of the direct sound has been either neglected for the design of the equalization filter [4], or iterative procedures have been utilized to compute the equalization filter [3]. Furthermore, a non-iterative approximate all-pass design has been proposed to obtain an equalization filter [8]. While this incorporates the direct sound and processing delay, it does not allow to include the electro-acoustic characteristics of the device. Therefore, in this paper we a propose non-iterative design of the equalization filter that takes into account the direct sound, individually measured transfer functions of the device and the processing delay. The proposed equalization filter design method requires a set of transfer functions that can and should be measured in-situ for each individual person [3].

The paper is structured as follows: In Section 2, the acoustic scenario is introduced and problems occurring in designing an appropriate equalization filter are analysed. In Section 3, the filter design methods based on a frequency-domain least squares cost function with various extensions are proposed. Sections 4 and 5 describe verification simulations, as well as results for the introduced filter design methods. Finally, the findings are summarized in Section 6.

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2 Problem Statement

Consider the acoustic scenario shown in Figure 1. The loudspeaker at the bottom represents a calibration sound source that is under control for measuring the relevant transfer functions.

In the open ear case shown on the left side of Figure 1, the signal at the eardrum is the source signal filtered by the acoustic transfer function to the eardrum of the open ear $D_o(\omega)$ at radial frequency ω. For the aided case as shown on the right side of Figure 1, the signal at the eardrum is the superposition of a source signal filtered by the acoustic transfer function of the semi-occluded ear $D_c(\omega)$ and the source signal filtered by the transfer function to the device microphone $D_m(\omega)$, the equalization filter $G_{EQ}(\omega)$ and the transfer function between the device loudspeaker and the eardrum $D_l(\omega)$. For convenience, we assume that all processing delay is included in the transfer function of the device loudspeaker $D_l(\omega)$. The transfer function of the complete system in the aided case (neglecting feedback) is then given by

$$D_{aided}(\omega) = D_m(\omega)G_{EQ}(\omega)D_l(\omega) + D_c(\omega). \qquad (1)$$

While for both cases the sound cannot be measured directly at the eardrum, for the aided cases methods exists that employ a microphone at the inner face of the device to obtain an estimate

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of the sound pressure at the eardrum [9, 10]. Therefore, in the following we assume availability of the signal at (or the transfer function to) the eardrum in the aided case.

Acoustic transparency means that the transfer function to the eardrum is equivalent in the open and the aided case, i.e.,

$$D_{aided}(\omega) = D_o(\omega). \tag{2}$$

Note that in practice, the open ear transfer function $D_o(\omega)$ is unknown and needs to be estimated. Such an estimate $\hat{D}_o(\omega)$ can be obtained, e.g., by applying an appropriate transformation function $G_T(\omega)$ to the microphone transfer function [11], i.e.,

$$\hat{D}_o(\omega) = G_T(\omega) D_m(\omega). \tag{3}$$

In the following, this approximation is referred to as target transfer function. The optimal equalization $G_{EQ}^{(opt)}(\omega)$ is then obtained by requiring

$$\begin{aligned} \hat{D}_o(\omega) &= D_{aided}(\omega) \\ &= D_m(\omega) G_{EQ}^{(opt)}(\omega) D_l(\omega) + D_c(\omega), \end{aligned} \tag{4}$$

and solving for the equalization filter, yielding

$$\begin{aligned} G_{EQ}^{(opt)}(\omega) &= \frac{\hat{D}_o(\omega) - D_c(\omega)}{D_m(\omega) D_l(\omega)} \\ &= \frac{1}{D_l(\omega)} \left(G_T(\omega) - \frac{D_c(\omega)}{D_m(\omega)} \right). \end{aligned} \tag{5}$$

Since $D_l(\omega)$ contains a frequency-independent group delay with respect to the other transfer functions, $G_{EQ}^{(opt)}$ is generally acausal, which is not realizable in practice. Also, exact inversion of D_m and D_l might not be possible, since deep notches can occur there. Hence, a filter design method is needed that allows the computation of a realizable (causal) equalization filter while minimizing the differences between the aided transfer function and the open ear transfer function.

3 Filter Design

In this section we present the proposed equalization filter design using a least-squares optimization procedure to obtain a causal filter. While we optimize the time-domain filter coefficients, it is practical to specify the desired transfer functions in the frequency-domain. Therefore, in Section 3.1 we first introduce the frequency-domain representation of the time-domain filter coefficients. In Section 3.2 we formulate the computation of the filter coefficients as a frequency-domain least-squares optimization problem. In Section 3.3 we introduce an acausality management and in Section 3.4 we propose to incorporate a frequency-dependent regularization to reduce comb filtering effects.

3.1 Frequency-Domain Optimization of Time-Domain Filter Coefficients

Since the target for acoustic transparency is defined as a transfer function, the equalization filter is computed based on a frequency-domain cost function. However, for future implementation on a hearing device, the time domain filter coefficients $\mathbf{g}_{EQ}$ (vector of length N_T) or spectral coefficients decoupled from the spectral resolution of the transfer functions are required. Making the desired length of the time-domain filter N_T independent of the Discrete Fourier Transform (DFT) length $N_F \geq N_T$ used to calculate the transfer functions, we write

$$\mathbf{G}_{EQ} = \underline{\underline{\mathbf{F}}} \mathbf{g}_{EQ} \tag{6}$$

$$\text{with} \quad \underline{\underline{\mathbf{F}}} = \underline{\underline{\mathfrak{F}}}^{(N_F \times N_F)} \begin{bmatrix} \underline{\underline{\mathbf{I}}}^{(N_T \times N_T)} \\ \underline{\underline{\mathbf{O}}}^{(N_F - N_T \times N_T)} \end{bmatrix}. \tag{7}$$

$\underline{\underline{\mathfrak{F}}}$ is the DFT matrix, $\underline{\underline{\mathbf{I}}}$ and Identity matrix and $\underline{\underline{\mathbf{O}}}$ a matrix containing only zeros, and $\mathbf{G}_{EQ}$ is the discrete frequency response of the equalization filter.

3.2 Least-Squares Cost Function

The equalization filter should minimize the difference between the aided and open ear transfer function to the eardrum. Similarly to Eq. (4), we therefore define a least-squares cost function of the form

$$J_{LS}(\mathbf{g}_{EQ}) = ||(\underline{\underline{\mathbf{D}}}_m \underline{\underline{\mathbf{D}}}_l \mathbf{G}_{EQ} + \mathbf{D}_c) - \hat{\mathbf{D}}_o||_2^2, \tag{8}$$

where $\underline{\underline{\mathbf{D}}}_m$ and $\underline{\underline{\mathbf{D}}}_l$ are diagonal matrices containing the DFT coefficients of $D_m(\omega), D_l(\omega)$ respectively; $\mathbf{D}_c$ and $\hat{\mathbf{D}}_o$ are according vectors. The optimum with respect to $\mathbf{g}_{EQ}$ is given by

$$\mathbf{g}_{EQ}^{(LS)} = \left(\underline{\underline{\mathbf{A}}}^H \underline{\underline{\mathbf{A}}} \right)^{-1} \underline{\underline{\mathbf{A}}}^H (\hat{\mathbf{D}}_o - \mathbf{D}_c), \tag{9}$$

where

$$\underline{\underline{\mathbf{A}}} = \underline{\underline{\mathbf{D}}}_m \underline{\underline{\mathbf{D}}}_l \underline{\underline{\mathbf{F}}}, \tag{10}$$

and $(\cdot)^H$ denotes the hermitian transpose of a matrix.

3.3 Acausality Management

To avoid potential acausality problems, the filter $\mathbf{g}_{EQ}$ is forced to be shifted in time by writing

$$\tilde{\mathbf{G}}_{EQ} = \underline{\underline{\mathbf{z}}}^D \underline{\underline{\mathbf{F}}} \mathbf{g}_{EQ}. \tag{11}$$

There, $\underline{\underline{\mathbf{z}}}^D$ denotes a diagonal matrix whose elements are the phase coefficients corresponding to a negative shift in time by D samples. D is chosen to be the processing delay, extended by some additional samples that allow for small acausalities in the filter design. The optimization for $\mathbf{g}_{EQ}$ is performed analogous to the previous section, yielding

$$\mathbf{g}_{EQ}^{(LSD)} = \left(\underline{\underline{\mathbf{A}}}_D^H \underline{\underline{\mathbf{A}}}_D \right)^{-1} \underline{\underline{\mathbf{A}}}_D^H (\hat{\mathbf{D}}_o - \mathbf{D}_c), \tag{12}$$

where

$$\underline{\underline{\mathbf{A}}}_D = \underline{\underline{\mathbf{D}}}_m \underline{\underline{\mathbf{D}}}_l \underline{\underline{\mathbf{z}}}^D \underline{\underline{\mathbf{F}}}. \tag{13}$$

3.4 Regularization

Comb filter effects are most pronounced in frequency regions where the direct sound is not attenuated significantly with respect to the target transfer function [3]. To include this observation in the design process, a frequency dependent regularization imposing an additional cost for these frequency regions is included in the cost function, similarly to [12]

$$J_{LSR}(\mathbf{g}_{EQ}) = ||(\underline{\underline{\mathbf{D}}}_m \underline{\underline{\mathbf{D}}}_l \tilde{\mathbf{G}}_{EQ} + \mathbf{D}_c) - \hat{\mathbf{D}}_o||_2^2 + \mu ||\underline{\underline{\mathbf{V}}} \tilde{\mathbf{G}}_{EQ}||_2^2, \tag{14}$$

where $\underline{\underline{\mathbf{V}}}$ is a real-valued diagonal matrix whose diagonal entries are spectral regularization weights for each frequency bin of $\tilde{\mathbf{G}}_{EQ}$, and μ is the regularization parameter. The filter optimizing Eq. (14) is given by

$$\mathbf{g}_{EQ}^{(LSDR)} = \left(\underline{\underline{\mathbf{A}}}_D^H \underline{\underline{\mathbf{A}}}_D + \mu \underline{\underline{\tilde{\mathbf{V}}}}^H \underline{\underline{\tilde{\mathbf{V}}}} \right)^{-1} \underline{\underline{\mathbf{A}}}_D^H (\hat{\mathbf{D}}_o - \mathbf{D}_c), \tag{15}$$

with

$$\underline{\underline{\tilde{\mathbf{V}}}} = \underline{\underline{\mathbf{V}}} \underline{\underline{\mathbf{z}}}^D \underline{\underline{\mathbf{F}}}. \tag{16}$$

We compute the frequency-dependent regularization weight $\mathbf{V}$ based on the level relation of the direct sound $\mathbf{D}_c$ and the target transfer functions $\hat{\mathbf{D}}_o$ in each DFT bin k

$$V[k] = \sum_{k'=0}^{N_F - 1} S_k[k'] \min\left(1, \left| \frac{\hat{D}_o[k']}{D_c[k']} \right| \right)^{N_v}. \tag{17}$$

The constrained relation of target and direct sound transfer function is expanded with an exponent N_v and then smoothed by applying a smoothing vector $\mathbf{S}_k$ [13]. Here, smoothing across 1/6 octave with a rectangular smoothing window was performed and the expansion N_v was set to 5.

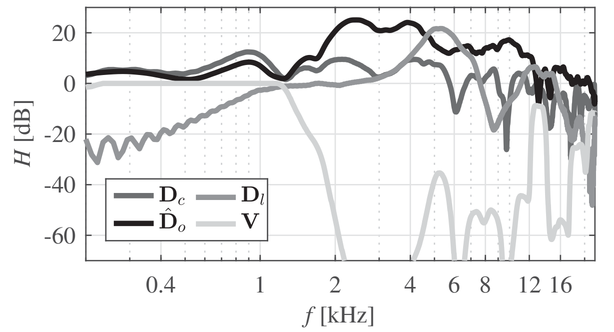

Figure 2: Acoustic transfer functions utilized for the simulations. The regularization weight $\mathbf{V}$ was calculated according to Eq. (17)

4 Measurements & Simulations

The necessary acoustic transfer functions have been measured in a human subject in a free field setup as described in more detail in [10, 14]. The hearing device as described by Denk et al. [3] was utilized. It is an individual earmould including a vent, 3 microphones and 2 loudspeakers, in an assembly corresponding well to Figure 1. Here, as in [3] only one loudspeaker located at the inner face of the device and one microphone (located at the back of the concha) were utilized. Measurements at the eardrum were performed using a probe tube microphone. For the present simulations, the influence of microphone sensitivities were equalized out, i.e., purely acoustic transfer functions are utilized. For all simulations, a single sound source and frontal sound incidence at 1 m distance were considered. Contributions of feedback to the hearing device microphone were neglected.

The utilized transfer functions are shown in Figure 2. Below about 1.5 kHz, the device does not attenuate external sounds. Between 500 Hz and 1.3 kHz, the transfer function to the eardrum of the occluded ear canal is larger than the open ear transfer function to the eardrum, which is presumably caused by a Helmholtz resonance of the residual ear canal and the vent. Above 2 kHz, the device attenuates external sounds by about 20 dB on average with respect to the open ear transfer function, however the behaviour is highly frequency-dependent.

The processing delay of the device was set to 6 ms and implemented as a time shift applied to D_l. A sampling rate of 48 kHz was utilized, and spectral analysis was performed with a DFT length of $N_F = 4096$ samples. Using the transfer functions and other parameters, the equalization filters and corresponding aided transfer functions were computed for the approaches described in the previous section.

4.1 Error metric

As an error metric between two transfer functions $\mathbf{H}_1$ and $\mathbf{H}_2$, a perceptually motivated auditory spectral distance was computed. Therefore, the mean difference between amplitudes in dB is averaged with a weight in a frequency range bounded by the bins k_1 and k_2

$$\Delta H_{\text{Aud}} = \sum_{k=k_1}^{k_2} W[k] |10\log_{10}(|H_1[k]|^2) - 10\log_{10}(|H_2[k]|^2)|. \tag{18}$$

$W[k]$ is a frequency-dependent weight, which was chosen as the inverse of the ERB-bandwidth depending on frequency to counteract over-representation of high frequencies [15]. It is normalized such that

$$\sum_{k=k_1}^{k_2} W[k] = 1. \tag{19}$$

Here, the frequency range in which this error is computed is constrained between 200 Hz and 16 kHz.

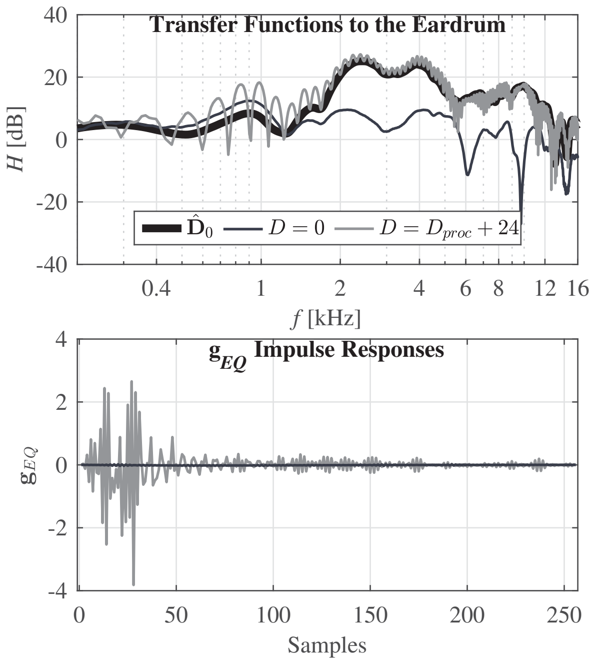

Figure 3: Top panel: Target and aided transfer functions, $\mathbf{g}_{EQ}$ computed using Eq. (9) and Eq. (12), without ($D = 0$) and with ($D = D_{proc} + 24$, i.e. 0.5 ms excess delay) including acausality management; filter length $N_T = 256$. Bottom panel: corresponding equalization filter coefficients $\mathbf{g}_{EQ}$.

5 Results and Discussion

5.1 Influence of Acausality Management

Figure 3 shows the target transfer function $\hat{\mathbf{D}}_0$ together with the aided transfer function and equalization filter impulse responses obtained by Eq. (9) and Eq. (12), i.e., not including and including the acausality management. For the acausality management, D was chosen to be the processing delay D_{proc} + 24 samples excess delay, i.e., an additional 0.5 ms. The filter length N_T was 256 samples.

Without the acausality management, the filter coefficients are close to 0, i.e., no sensible filter is computed. In consequence, the aided transfer function is almost equal to the occluded transfer function $\mathbf{D}_c$. Further simulations showed that less excess delay than the utilized 0.5 ms resulted in a poorer performance, while more excess delay did not result in a further improvement. Apparently, besides exploiting knowledge about the processing delay in the optimization process, it is required to allow for some acausality in the filter design to achieve good performance. The acausal taps probably support the partial equalization of the non-minimum phase system.

5.2 Influence of Regularization

Figure 4 shows the target and aided transfer functions as well as impulse- and frequency responses of $\mathbf{g}_{EQ}$, where $\mathbf{g}_{EQ}$ was calculated according to Eq. (14), i.e., including the acausality management and variable regularization. The aided transfer functions are shown for 3 different regularization weights $\mu = \{0, 10^{2.5}, 10^8\}$, and as in the previous section using a filter length $N_T = 256$ and D corresponding to the processing delay D_{proc} + 0.5 ms.

The influence of the regularization is generally positive on the performance: Comb-filter effects are significantly reduced as compared to the same setting where no regularization is applied (compare result for $\mu = 0$ against $\mu = 10^{2.5}$). This is because $\mathbf{G}_{EQ}$ reduces the hearing device output in frequency regions where no output is needed, i.e., where the direct sound transfer function $\mathbf{D}_c$ already provides sufficient level (c.f. Figure 2). Also,

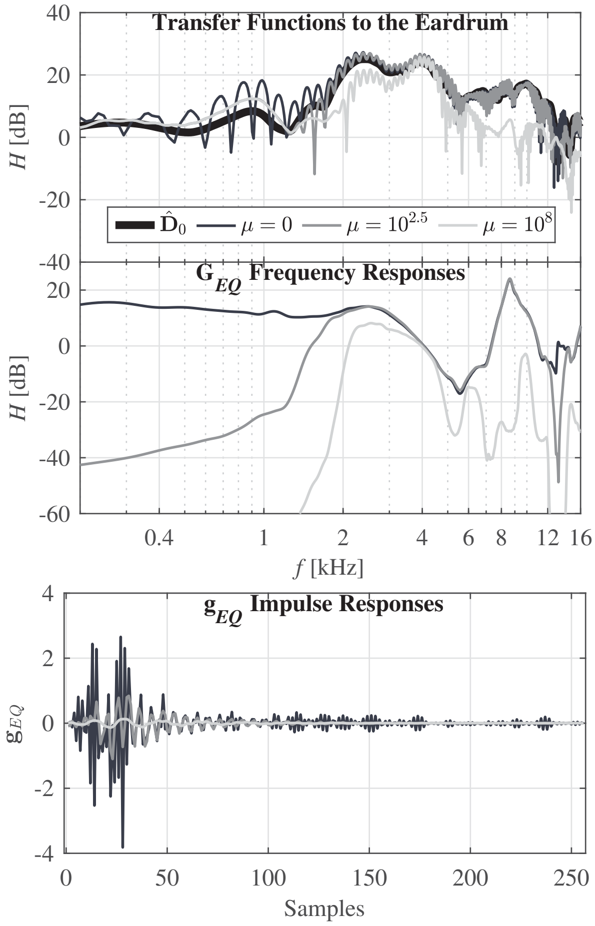

Figure 4: Top panel: Target and achieved aided transfer functions, $\mathbf{g}_{EQ}$ computed using Eq. (15) with different regularization parameters μ, filter length of $N_T = 256$ and $D = D_{proc} + 24$, i.e., and excess delay of 0.5 ms, as in Figure 3. Middle Panel: Frequency response of the equalization filters. Bottom Panel: Corresponding time-domain filter coefficients $\mathbf{g}_{EQ}$.

the filter coefficients in time-domain look better behaved when a regularization is applied: Less high-frequency ringing artefacts are noted, and the level is generally lower, additionally resulting in a smaller energy consumption due to a smaller amplification.

However, very large regularization weights result in a poor performance. In Figure 4, it becomes clear that for μ as large as 10^8, the frequency response of the equalization filter $\mathbf{G}_{EQ}$ becomes too small in level to provide the desired compensation of attenuation that the device produces by partially occluding the ear.

Figure 5 shows the auditory spectral distances ΔH_{Aud} according to Eq. (18) for variable μ and different filter lengths N_T. Independent of the filter length, a U-shaped error curve over μ is observed. Apparently, there is a trade-off between imposing an additional cost between avoiding signal playback in frequency regions where this is unnecessary, and the regularization aspect of the cost function dominating the optimization: If μ is too small, comb filter effects are not avoided, but if μ is too large, the cost for energy in $\mathbf{G}_{EQ}$ dominates the optimization, resulting in aided transfer functions that are too low. This means that there is an optimal μ, which for the majority of the regarded filter lengths, lies between 10^2 and 10^3.

5.3 Influence of the Filter Length

While in the previous sections a fixed filter length was considered, in this section we compare the results for different filter lengths N_T. Again, consider Figure 5, which shows the achieved spectral

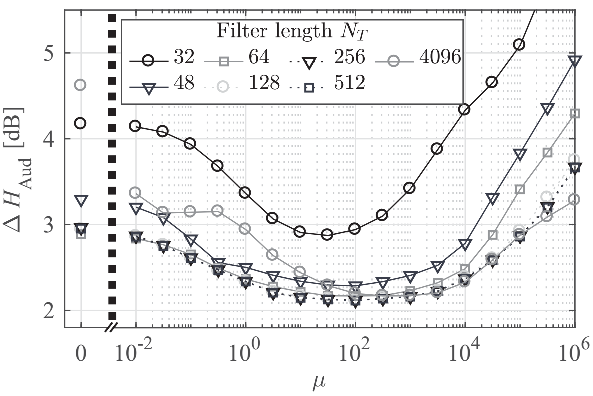

Figure 5: Auditory spectral distance between target and aided transfer functions with $\mathbf{g}_{EQ}$ computed using Eq. (15) depending on the regularization parameter μ, with D set to the processing delay + 24 samples (= 0.5 ms)

distance ΔH_{Aud} (c.f. Eq. 18) between the target and the aided transfer function. There, results for different filter lengths N_T and regularization parameters μ are compared.

The filter length has only small influence on ΔH_{Aud}, if a minimal length (about 64 samples) is exceeded. An auditory spectral difference between open and aided transfer function of 2.1 dB seems to be the lower limit for the given filter design and transfer functions. Probably, the observation that the direct sound transfer function $\mathbf{D}_c$ is larger than the target transfer function $\hat{\mathbf{D}}_0$ around 1 kHz contributes to a lower limit that deviates notably from 0. For very short filters with $N_T < 64$ samples, the residual spectral difference is larger, presumably due to a frequency resolution that is too poor. However, a very long filter ($N_T = 4096$, equal to DFT length) does not result in better performance – again, no cancellation of spectral ripples is achieved, although the spectral resolution of the filter would be sufficient to compensate the ripples by means of its amplitude.

It is also worth mentioning that the optimal regularization weight μ interacts with the filter length: The optimum μ increases with increasing filter length. This is understandable in a way that with very short filters, a highly over-determined set of equations is solved by Eq. (15), which by itself is somewhat equivalent to a regularization.

6 Conclusions

We presented an approach to design equalization filters for semi-open fit hearing devices with the aim to provide acoustic transparency. The approach is based on a frequency-domain least-squares cost function that takes into account individually measured transfer functions and the processing delay of the device, and has a closed-form solution for the time-domain filter coefficients. Furthermore, using the presented approach it is possible to decouple the spectral analysis length from the desired filter length.

Within this design approach, the results showed that it is critical to include an acausality management of the filters, where knowledge about the processing delay can be explicitly exploited. Furthermore, frequency-dependent regularization of the energy contained in the equalization filter resulted in a reduction of comb-filtering effects. The regularization parameters were computed automatically from the occurring acoustic transfer functions, which makes the regularization approach easily applicable. In the present work, filter lengths as short as 64 samples or 1.5 ms were sufficient to achieve the best possible performance.

In conclusion, using a least-squares design approach that takes the processing delay and automatic regularization into account, it is possible to compute individualized equalization filters for acoustically transparent hearing devices that produce only little comb filter artefacts.

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