Calibration of a mobile hearing aid prototype and its validation: Towards transparent listening with commodity hardware

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Introduction

Mobile hearing aid prototypes running open-source software allow for immediate testing under field conditions, and easy management and distribution of algorithms. Besides low latency and acoustic feedback, an accurate calibration of the device is important for the validity of any experiments. Ideally, the listener would not perceive any difference between listening with and without the device, especially in level and tone colour (transparent listening) [1].

This work presents the calibration of a mobile hearing aid prototype built with commodity hardware parts [2]. Using impulse responses measured with a dummy head in a diffuse field condition, compensation filters were determined to achieve the same levels at the eardrum for open ear and signals processed with the hearing aid prototype. Diffuse field equalization is used instead of free field equalization as recommended by [3]. The measurements were conducted for five pairs of binaural in-ear headphones, for the purpose of estimating a generic calibration from the averaged impulse responses. A generic calibration would allow to use the calibrated hearing aid prototype setup "out-of-the-box", for any user that wants to build a prototype based on the documentation provided in the wiki [2]. Such a pre-configured setup enables interested individuals and institutions far beyond the usual target group to operate a research hearing device. It may serve as an infrastructure to rapidly share and evaluate new signal processing strategies.

Methods

Hearing aid prototype

The mobile hearing aid prototype project [2] offers open-source code and documentation to allow everyone to build a prototype hearing aid with commodity hardware components. A pre-configured SD card image (current version 2.0) is available which is appropriate for the direct use of the prototype. The software openMHA [4] is used for the hearing aid signal processing. Hence, it gives its user the possibility to experience the effect of hearing aid processing on sound perception, and to test the standard algorithms provided with openMHA or even implement new ones.

The in-ear headphones (Roland CS-10EM) used in the prototype are equipped with microphone and receiver

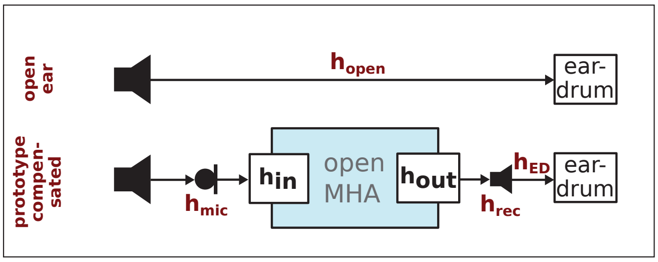

Figure 1: Signal paths from sound field (free field) to the eardrum of the dummy head in the virtual reality lab; for open ear canal (top) and for closed ear canal with compensation filters (bottom). In the aided case, the signal is recorded by the microphone, filtered with the input filter, processed with openMHA (inactive during the measurements of impulse responses), filtered with the output filter, and played back by the receiver.

per ear. The sound field is recorded with the binaural microphones, processed by openMHA, and presented to the listener via the receivers of the headphones. Thereby, the levels that would be present at the listener's eardrum when the ear canal was open are potentially altered.

Calculation of compensation filters for frequency-dependent calibration

Figure 1 shows a block diagram of the signal paths with and without using the hearing aid prototype. The sensitivity of the microphone needs to be equalized for correct measurement of frequency-dependent levels. Moreover, the output of the receiver needs to be corrected such that the desired open-ear levels are obtained at the eardrum after processing.

For the purpose of equalizing the microphones, FIR filters need to be determined that are able to compensate for the alterations. Figure 1 (bottom) illustrates the compensated signal path. The signal picked up by the microphone is filtered with the to-be-determined input filter $\mathbf{h_{in}}$ which aims at equalizing the microphone sensitivity in diffuse field. After the openMHA processing (not active during the calibration measurements), the processed signal is filtered with the output filter $\mathbf{h_{out}}$ which aims at equalizing the receiver sensitivity and providing the gain necessary for obtaining the correct levels at the eardrum.

The following impulse responses were measured for five pairs of headphones (left and right) using a dummy head (G.R.A.S. KEMAR type 45BB-12) in approximate diffuse-field conditions using 47 loudspeakers in the virtual reality lab of the University of Oldenburg:

- $\mathbf{h_{mic}}$: playback: loudspeakers, diffuse field; recording: microphone of the headphones (positioned in free-field, no dummy head)
- $\mathbf{h_{rec}}$: playback: receiver of the headphones; recording: microphone at the eardrum of dummy head
- $\mathbf{h_{open}}$: playback: loudspeakers, diffuse field; recording: microphone at the eardrum of dummy head
- $\mathbf{h_{ED}}$: playback: loudspeakers, diffuse field; recording: microphone at the eardrum of dummy head in aided condition

With these impulse responses available, the desired FIR inversion filters were estimated using the procedure by Kirkeby & Nelson [5]:

An inversion filter $\mathbf{h}$ is calculated by solving a linear equation system for which the solution for $\mathbf{h}$ is given by Equation 1 [5]. Here, $\mathbf{C}$ is a convolution matrix for the impulse response $\mathbf{c}$ which is to be inverted, convolution of $\mathbf{c}$ with $\mathbf{h}$ should lead to the target signal $\mathbf{a}$. $\mathbf{B}$ is a convolution matrix for the regularization filter $\mathbf{b}$ which is a high-pass filter (cut-off frequency 16 kHz) applied for the purpose of avoiding an "excessive boost of unwanted frequencies", hence applying regularization only at high frequencies. A trade-off between accuracy (minimal performance error) and the energy at high frequencies can be controlled by the regularization parameter β.

$$\mathbf{h} = [\mathbf{C}^T\mathbf{C} + \beta\mathbf{B}^T\mathbf{B}]^{-1} \cdot \mathbf{C}^T\mathbf{a} \tag{1}$$

Additionally, in the procedure of by Kirkeby & Nelson [5] the parameters filter delay D and filter length N can be defined. These parameters were optimized for obtaining a minimal error between reproduced signal and target signal.

Input filter
The inversion filter $\mathbf{h_{in}}$ was determined by inverting the impulse response $\mathbf{h_{mic}}$ (averaged over loudspeakers for diffuse field equalization and over headphones) using Equation 1, with a delta impulse as target signal $\mathbf{a}$. After the deconvolution, a flat frequency response should be obtained after processing the microphone output.

Output filter
The inversion filter $\mathbf{h_{out}}$ was determined using the impulse response of the receiver $\mathbf{h_{rec}}$ as $\mathbf{c}$, but with a frequency-dependent target response. $\mathbf{h_{out}}$ was designed such that $\mathbf{h_{rec}} * \mathbf{h_{ED}}$ resembled the impulse response $\mathbf{h_{open}}$. Therefore, the target gain was determined by deconvolving the impulse responses $\mathbf{h_{open}}$ and $\mathbf{h_{ED}}$ (division in frequency domain) because the output filter is applied before playback with the receiver, and the signal path inside the ear canal is the same for open ear and prototype-compensated condition.

Results

Input filter
Input filters $\mathbf{h_{in}}$ were determined for all parameter combinations of

- regularization parameter $\beta = \{1, 0.5, 0.25, 0.1, 0.05, 0.02, 0.01, 0.005, 0.002\}$,
- filter delay $D = \{-4 : 1 : 20\}$ samples, and
- filter length $N = \{4, 6, 8, 12, 16, 24, 32, 48, 64\}$ samples.

Figure 2 shows the weighted average deviation for input filters $\mathbf{h_{in}}$, calculated with regularization parameter $\beta = 0.25$ (see below for further explanation on the choice of β), and all combinations of filter delay D and filter length N. The weighted average deviation describes the difference between the calculated generic filters using the respective parameter combination, and a "perfect" inversion filter. The frequency-dependent deviations are summed using a $1/f$ weighting. $N = 16$ and $D = 0$ were chosen due to the considerations that N should be as low as possible given a small error (computing time) and for different delays D the error was comparable.

Figure 3 shows the deviation of the different headphones over frequency for the chosen parameters, normalized to 1 kHz where 0 dB deviation is defined as reference. Only small deviation of 1-2 dB occur for frequencies up to 6 kHz. Differences between the headphones are only present for high frequencies.

Figure 4 shows the magnitude of the input filter $\mathbf{h_{in}}$ for the chosen parameter combination $\beta = 0.25$, $N = 16$, and $D = 0$ (red), which is applied in the hearing aid prototype [2] starting from SD image v1.2. For demonstration purposes, filters for different regularization parameters β are displayed in green and blue - for small β, frequencies above 16 kHz are amplified, for $\beta = 1$ these frequencies are attenuated. The determined filter with $\beta = 0.25$ provides a flat frequency response over a wide frequency range with a moderate attenuation of high frequencies.

Output filter
Output filters $\mathbf{h_{out}}$ were determined for all combinations of

- regularization parameter $\beta = \{1, 0.5, 0.25, 0.1, 0.05, 0.02, 0.01, 0.005, 0.002\}$,
- filter delay $D = \{-4 : 1 : 20\}$ samples, and
- filter length $N = \{8, 12, 16, 24, 32, 48, 64, 96\}$ samples.

Figure 5 shows (analog to Figure 2) the weighted average deviation for output filters $\mathbf{h_{out}}$, calculated with regularization parameter $\beta = 0.01$, and all combinations of filter delay D and filter length N. With the choice of $N = 64$ and $D = 0$, a minimal weighted average deviation of about 3 dB could be obtained.

Figure 6 shows the deviation of the different headphones in the same representation as Figure 3. Here, the impulse responses for left and right were averaged for each pair

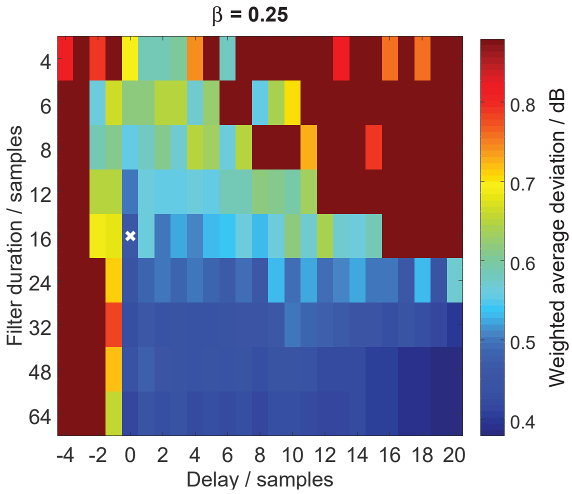

Figure 2: Weighted average deviation in dB for input filters determined using different combinations of filter duration N and filter delay D, with the selected regularization parameter $\beta = 0.25$.

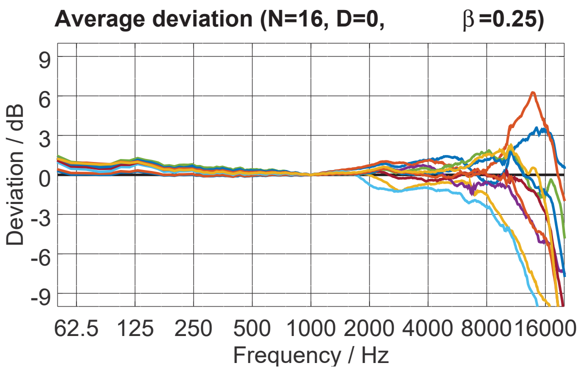

Figure 3: Spectral energy deviation of compensated impulse responses of the microphones relative to 1 kHz, for different headphones.

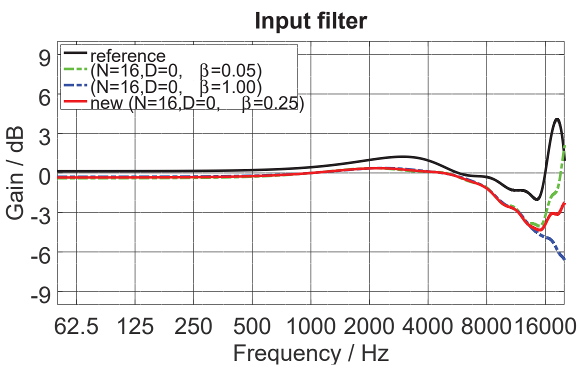

Figure 4: Transfer function of input filter selected to use in the hearing aid prototype (red). Additionally, the previously used filter is shown (black), and the selected new filters with different β parameter (green and blue).

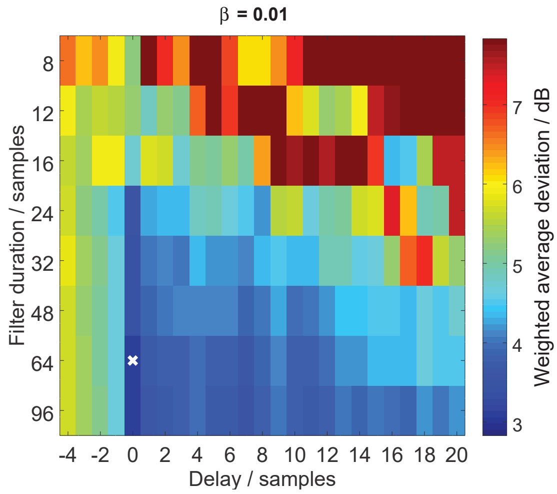

Figure 5: Weighted average deviation in dB for output filters determined using different combinations of filter duration N and filter delay D, with the selected regularization parameter $\beta = 0.01$.

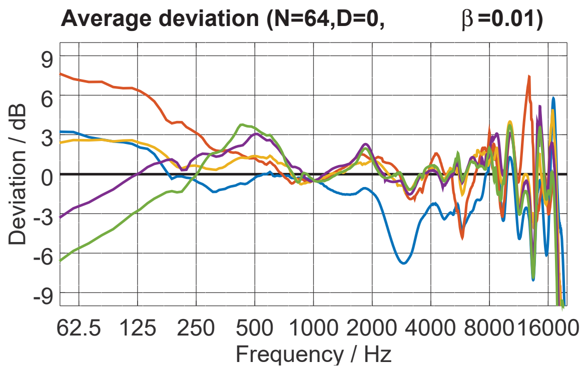

Figure 6: Spectral energy deviation of compensated impulse responses of the receivers relative to the average target gain and relative to 1 kHz, for different headphones (average between left and right).

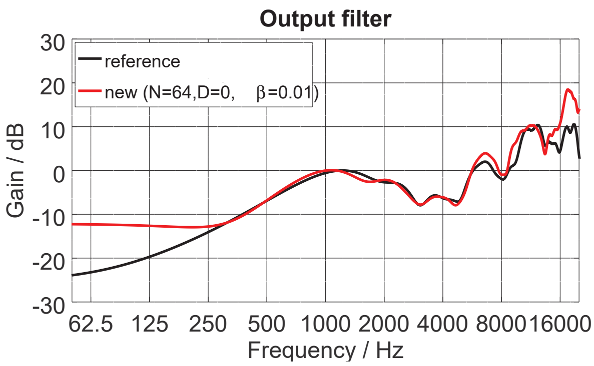

Figure 7: Transfer function of output filter selected to use in the hearing aid prototype (red). Additionally, the previously used filter is shown (black).

of headphones. Higher deviations between headphones as for the input filters occur.

Figure 7 shows the magnitude of the output filter $\mathbf{h_{out}}$ for the chosen parameter combination $\beta = 0.01$, $N = 64$, and $D = 0$ (red), which is applied in the hearing aid prototype [2] starting from SD image v1.2. The frequency-specific gain leads to a plausible hearing impression when testing the calibration applied in the prototype. Compared to a previous version of the determined filter (reference, black, used until SD image v1.1), low frequencies are less attenuated.

Discussion and Conclusions

Relatively short inversion filters with durations of 16 and 64 samples at a sample frequency of 48 kHz were found to be sufficient to calibrate the microphones and receivers of five pairs of Roland CS-10EM binaural headphones, respectively. These filters already take into account the relative transfer function from the microphone to the eardrum which aims at providing a transparent listening experience. While the quality was not assessed quantitatively, the subjective impression of the authors was that the calibration sounds approximately plausible and realistic, for example regarding tone colour.

A diffuse field equalization was applied by averaging over impulse responses of the loudspeakers of the virtual reality lab, which has the advantage of providing a better binaural perception than a free field calibration only accounting for a single sound source from frontal direction [3].

The generic calibration using dummy head measurements (no individual subjects) allows every potential user of the prototype to use the calibrated setup without needing access to a virtual reality lab where individual measurements of impulse responses could be conducted; and less effort is needed before starting to use the prototype. Nevertheless, listening experiments would be needed to verify if the generic calibration is sufficient for each individual listener.

Assuming that the calibration is valid within an error margin of about 3 dB in the range from 250 to 8000 Hz for input and output, the prototype can be used for calibrated binaural recordings as well as for measurements of the individual hearing thresholds.

Binaural recordings can be gathered with the prototype in any realistic situation. The self-fitting GUI described in [6] includes some examplary recordings. The recordings can be used to evaluate individual hearing aid fittings in realistic conditions. Using openMHA [4], different hearing aid algorithms can be applied on the prototype. The self-fitting GUI allows to try out different dynamic compression algorithms.

Hearing thresholds can be measured on the prototype using the single-interval adjustment-matrix (SIAM) procedure [7], since the Raspberry Pi 3 SOC is powerful enough to run GNU/Octave which can be easily used to implement psychoacoustic measurements. The SIAM procedure is already implemented on the prototype [8]. The measurement of hearing thresholds could also be used to verify the calibration. For individual listeners, the thresholds measured for open ear and the compensated thresholds measured with the prototype can be compared, when both are measured in the same conditions, e.g. both in the virtual reality lab.

References

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