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Perceptual loudness compensation for evaluation of personalized earbud equalization

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ABSTRACT

The ear canal geometries of individuals are very different, resulting in significant variations in SPL at the drum reference point (DRP). Knowledge of the personalized transfer function from a near-field microphone (NFM) in the earbud speaker tip to DRP allows personalized equalization (P.EQ). A method to perceptually compensate for loudness perception within the evaluation of different personalized equalization filters for earbuds has been developed. The method includes: measurements at NFM to estimate the personalized transfer function from the NFM point to DRP, calibration of the NFM microphone, the acquisition of the transfer function from the earbuds' speaker terminals to NFM, and the estimation of perceptual loudness in phones at DRP, when applying the different equalization filters. The loudness estimation was computed using the Moore-Glasberg method as implemented in ISO 532-2. The difference to DRP estimation was recursively adjusted using pink noise until the delta was within 0.1 dB. The corresponding gains were applied to the different conditions to be evaluated. A listening test was performed to evaluate three conditions using the described method for loudness compensation.

1 Introduction

Research has shown that more individualized equalization in headphones results in a better listening experiences [1, 2, 3]. A study described by Bezzola [4], has shown that the ear canal geometries of individuals are very different, resulting in significant variations in SPL at the drum reference point (DRP). This implies that earbuds which are tuned on a standard ear and mannequin would have different timber for the individual users.

Knowledge of the individual transfer function from the near-field microphone (NFM) to DRP allows for personalized equalization (P.EQ) of the sound pressure at the eardrum. In a recent study [5], an equalization method that uses measurements at NFM in the earbud's speaker tip, to estimate the personalized transfer function G, from NFM to DRP was introduced. Two main conditions were evaluated in [5], the P.EQ filter based on a transfer function G_K that was built on average measurements over KEMAR, and a P.EQ equalization

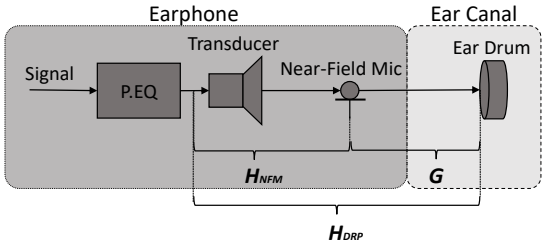


Fig. 1: Block diagram of the signal chain. P.EQ is the personalized equalization filter to be evaluated. H and G denote the transfer functions utilized through this study.

with G_p based on an estimation of the individual G transfer function.

In order to fairly compare only the spectral variations of the different equalization filters, a loudness level compensation needed to be performed between the test conditions. Much research has been produced about loudness perception, [6, 7, 8, 9, 10], this work is mainly based on the ISO-532, Part 2, [11].

On this work the evaluation of three different conditions: No-P.EQ, P.EQ with G_K, and P.EQ with G_p, is described. This paper also details the methodology to compensate for loudness, on the evaluation of different equalization conditions for an earbud.

2 Methods

The methodology for the computation of the transfer functions, equalization filters, and loudness compensation are detailed through this section.

2.1 Transfer function at NFM

The impulse response (IR) denoted as $\hat{h}_{NFM}$ measured at NFM was computed using a frequency domain adaptive filter (FDAF) algorithm, with the overlap-save method. This approach is efficient and effective for applications requiring precise modeling of acoustic transfer functions [12]. The filter coefficients of $\hat{h}_{NFM}$ where updated using a gradient descent algorithm, to minimize the error between the earbud's audio signals, at the speaker terminals, and at the NFM microphone. After acquiring the IR, the transfer function in dB, from the earbud's speaker terminals to NFM was obtained, by computing the FFT over $\hat{h}_{NFM}$.

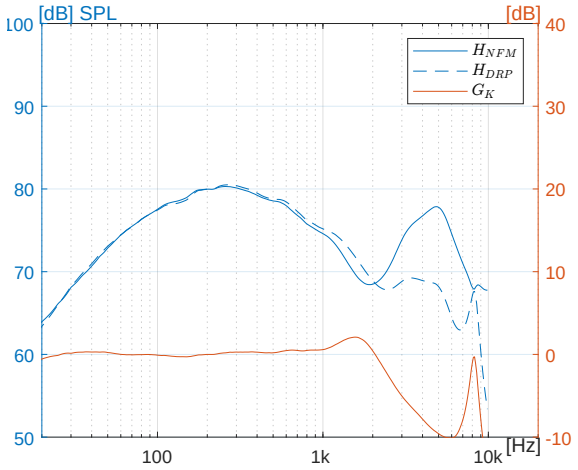


Fig. 2: KEMAR's transfer functions. H_{NFM} curve, H_{DRP} curve and G_K curve.

2.2 Transfer function G

The transfer function G in the frequency domain, corresponds to the relationship in dB from NFM to DRP, computed by

$$G = H_{DRP} - H_{NFM}, \tag{1}$$

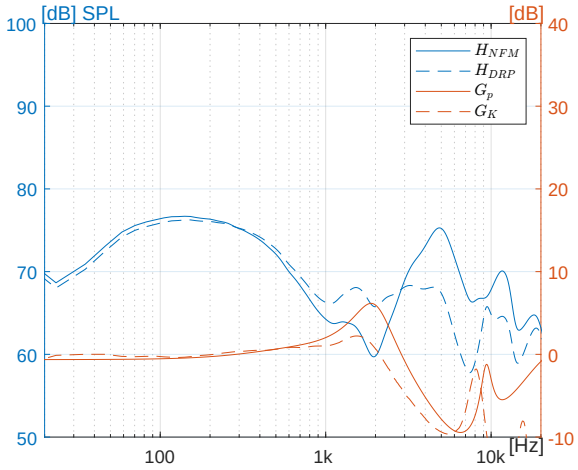


Fig. 3: Transfer functions, left ear's Listener ID 8. H_{NFM} curve, estimated H_{DRP} curve, personalized G_p curve, KEMAR's average G_K curve.

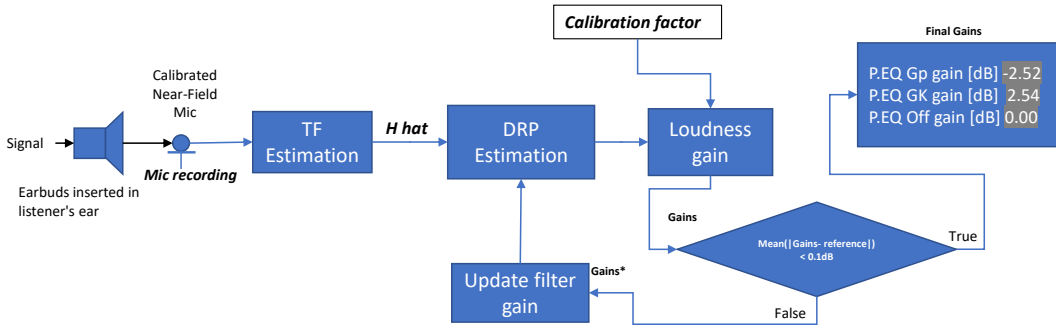


Fig. 4: Loudness compensation block diagram procedure.

where H_{DRP} denotes the transfer function from the speaker terminals to the eardrum, and H_{NFM} is the transfer function from the speaker terminals to NFM.

This function allows to estimate the pressure at the wearer's eardrum from a measurement at NFM, and reflects the physical characteristics of the wearer's ear canal. The estimated transfer function G_p was computed from a model based on FEM simulations in COMSOL, and from measurements on a KEMAR dummy head and torso [5]. On Figure 2, the transfer functions: H_{NFM}, H_{DRP} and G_K, of the earbud measured on KEMAR head an torso are shown. On Figure 7, the transfer functions: H_{NFM}, H_{DRP}, G_K, and the estimated G_p (Left ear, Listener ID 8), are presented.

2.3 Equalization filters

The target of the equalization filters is to achieve the same frequency response at each wearer's eardrum. This can be done by applying an equalization to the $NFM + G$ response as in

$$TG_{DRP} = P.EQ + (H_{NFM} + G), \tag{2}$$

where G is the estimated or measured transfer function from NFM to DRP, and TG_{DRP} is the predefined target at DRP, (see Figure 1). Then the equalization filter is calculated as

$$P.EQ = TG_{DRP} - (H_{NFM} + G). \tag{3}$$

In [5] two filters were evaluated: an IIR P.EQ based on a transfer function G_K, that was built on average measurements over KEMAR, and an FIR P.EQ, with Gp based on an estimation of the individual G transfer function. A comparison of three different conditions: No-P.EQ, P.EQ based on G_K, and P.EQ based on G_p were evaluated. The target curve TG_{DRP} at DRP is shown on Figure 7, Section 3.

2.4 Earbud setup

An earbud prototype similar to a commercially available earphone was utilized for this evaluation. The earbud included wire access to read the speaker and inner microphone signals, in order to measure the transfer function H_{NFM}. The speaker and microphone were connected to a portable PC through a full-duplex audio interface and controlled by a graphical user interface (GUI) described in Section 2.7. The total length of the IRs was 2048 samples, using a sampling frequncy Fs: 48 kHz. The frequency response H_{NFM} was obtained by computing an FFT, with a resolution of 11.7188 Hz.

2.5 Microphone calibration

In order to measure loudness precisely, the microphone of the earbud was calibrated. The procedure for the calibration was carried out as follows: first, the earbud was coupled to a G.R.A.S. 42AG Multifunction sound calibrator, secondly, a 1 kHz sinusoidal was played at 94 dB SPL through the sound calibrator and recorded by the NFM microphone. Finally the recording was evaluated by the Matlab's function `calibrateMicrophone` [13]. The function computes the calibration factor using:

$$CalibrationFactor = \frac{10^{((SPL-k)/20)}}{\mathrm{RMS}(x)}, \tag{4}$$

where x is the microphone recording passed through a A-weighting frequency filter. k is 1 Pascal relative to

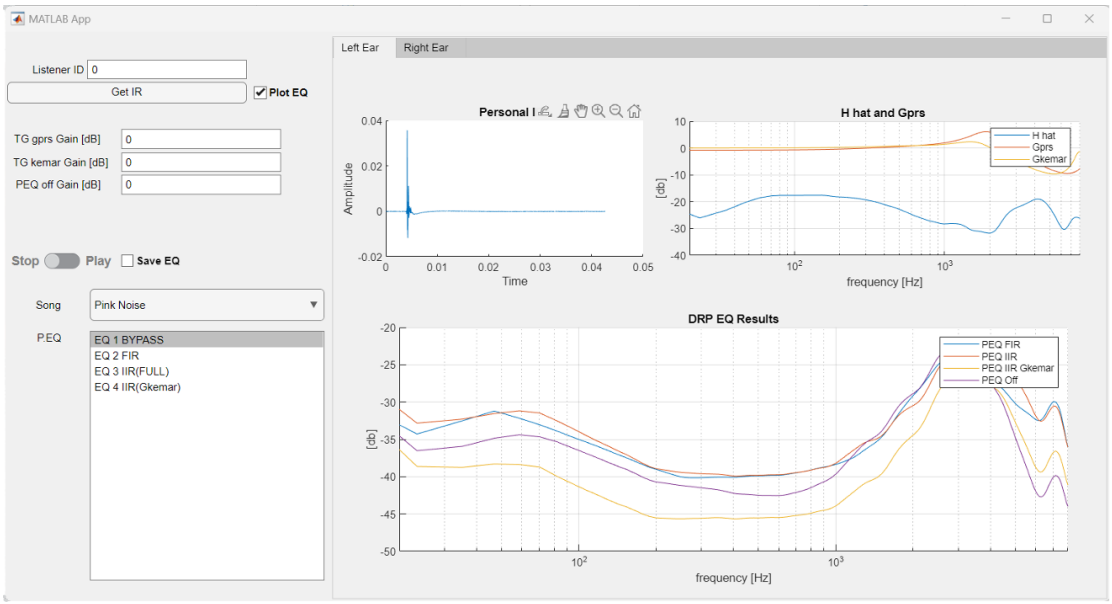


Fig. 5: GUI to acquire IRs, compute transfer functions, calculate filters and loudness compensation.

the *Pressure reference* calculated as

$$k = 20log_{10}\left(\frac{1}{Pressure\ reference}\right). \quad (5)$$

The function requires to input the sampling rate fs, the injected SPL level, and the *Pressure reference* of 20^{-6} Pa, for the calibration factor computation.

2.6 Perceptual Loudness estimation

The Moore-Glasberg method as implemented in ISO 532-2 [11] is limited to stationary sounds, and can be applied to tones, broadband noises and complex sounds with sharp line spectral components. The standard improves the precision of calculated loudness in the low frequency range, and allows for calculation of loudness under conditions where the sound differs at the two ears. The method accounts for the contours of equal loudness. The ISO 532-2 provides an algorithm for the calculation of monaural or binaural loudness: for sounds recorded using a single microphone, a head and torso simulator, or for sounds presented via earphones.

The function `acousticLoudness` in Matlab [13] allows to compute the perceptual loudness based on the ISO 532-2 standard model and permits different options of sound fields. We have used this function to estimate the perceived loudness for the different equalization filters. The input to the function was the estimated earphone frequency response at DRP, and it was configured to measure the *SoundField* at *eardrum*, using the ISO 532-2 model. The function required also the calibration factor, that was computed for the NFM microphone, and the recording sampling frequency. On Figure 4, the block diagram procedure for the loudness compensation is shown. The following steps are part of the algorithm to compensate for loudness perception:

1. Test signal (pink noise) was played back through the Earbuds inserted into listener's ears, then the pressure was recorded at NFM.
2. G_p transfer function from NFM to DRP is estimated based on H_{NFM} [5], and then the SPL at the eardrum reference point (DRP) is calculated.
3. Estimate of perceptual loudness in phones [11], then compare gain differences in phones to a desired reference loudness.
4. Add the gain difference in dB to the G_{DRP} estimation, then recursively compare the loudness difference and adjust the gains needed until the difference is minimized within 0.1 dB.
5. Output final gains to listening app.

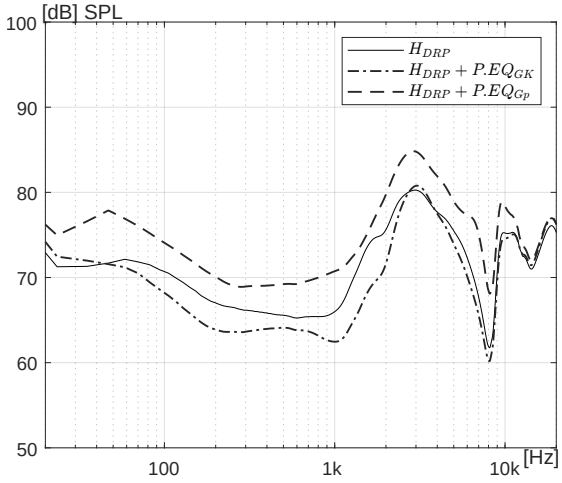


Fig. 6: Equalization result at DRP for Listener ID 8 without loudness compensation. H_{NFM} curve, $H_{NFM} + P.EQ_{GK}$ curve, and $H_{NFM} + P.EQ_{Gp}$ curve.

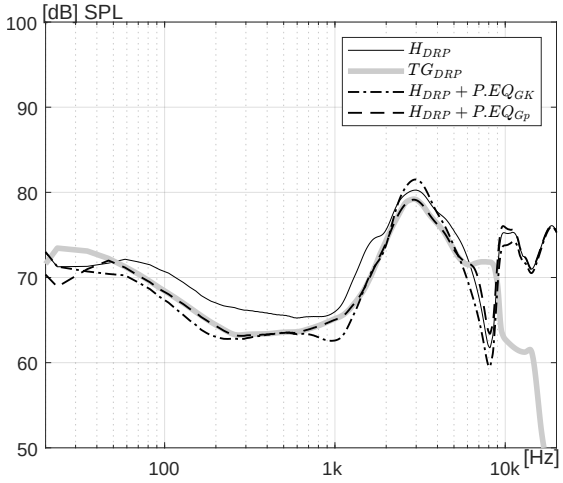


Fig. 7: Equalization result at DRP for Listener ID 8 with loudness compensation. H_{DRP} curve, target at DRP TG_{DRP} curve, $H_{DRP} + P.EQ_{GK}$ curve, $H_{DRP} + P.EQ_{Gp}$ curve.

2.7 Graphical user interface

A custom GUI was written in Matlab [13]. The GUI acquires the impulse responses $\hat{h}_{NFM}$, reads the output of the earbuds, computes the transfer functions Gp, generates the equalization filters, and performs the loudness adjustment for each condition. On Figure 5, the GUI utilized for this work is shown.

2.8 Listening test

A preference listening test was carried out to compare the three equalization conditions on insert earbuds: No-P.EQ, P.EQ with GK, and P.EQ with *Gp*. Four audio samples (music, speech and cinematic excerpts) were presented to the assessors through a customized test interface. The test conditions were obtained for each listener using the GUI. The corresponding filter coefficients were then calculated and applied in the test software. 16 participants completed the preference rating task using a 0-100 scale. The participants were encouraged to adjust the positioning of the earbuds for the most comfortable fit.

3 Results

In this section, the results on left ear's Listener ID 8, as well as the results of the listening test are presented.

The results of the equalization filters at DRP without loudness compensation are presented in Figure 6. Figure 7 shows the results of the equalization filters at DRP with loudness compensation are shown. The preference score of the listening test is presented in Figure 8.

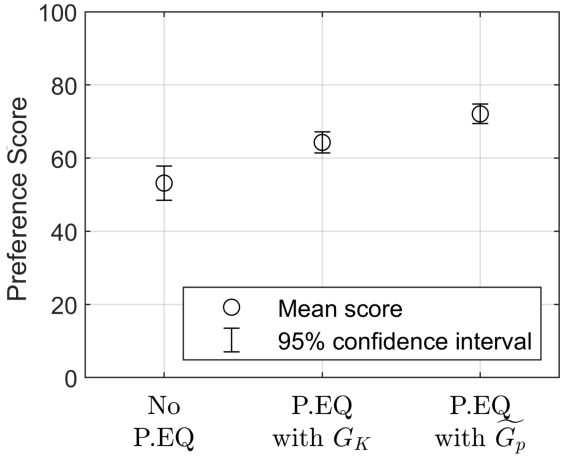


Fig. 8: Listening tests results adopted from [5]. showing the mean preference scores, and their 95% confidence intervals.

4 Discussion

Results show that both conditions with equalization were rated significantly higher than the No-P.EQ condition, and that the equalization based on the personalized G_p was rated significantly higher on average than the filter based on an the average G_K on KEMAR. As observed in Figure 7, on Section 3, the resulting frequency response curves matched closer the equalization target presented in the same Figure.

On preliminary listening tests, the score preferences where dominated by the equalization filter that had more overall energy output, therefore initially the level compensation was performed at the NFM point, using just the RMS level output measurement with pink noise. This procedure did not produce satisfactory results and the assessors reported that the different conditions were not level matched. When applying the loudness compensation the preference shown a more consistent result making a fair comparison between the evaluated conditions. The assessors also reported a more even perception between conditions. The main idea of the loudness compensation is to focus only on the spectral changes of the different conditions rather than the predominant level perception.

5 Summary

On this work, a loudness compensation method was applied to the evaluation of three different conditions for earbud equalization. The method was applied to measurements at NFM, to estimate the frequency response at DRP. The perceived loudness in phones was estimated at DRP for three conditions: no equalization, and two different equalization filters.

The loudness estimation was computed using the Moore-Glasberg method as implemented in ISO 532-2. The difference in dB to the DRP estimation was recursively adjusted, using pink noise until the delta was within 0.1 dB. The corresponding gains were applied to the three filter conditions.

A listening test was performed to evaluate the conditions, using the described method for loudness compensation. The listening test shown that, the equalization filter, based on the personalized G_p transfer function was preferred over the equalization based on a measured average G on KEMAR head and torso ear simulator, and over the no equalization condition.

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