Aalto University
School of Electrical
Engineering

Headphone Equalization Applications

Seminar Lecture

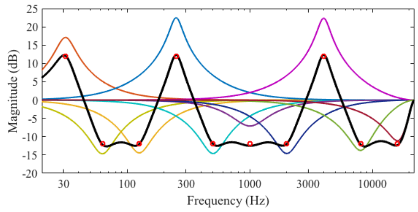
18.3.2019

Juho Liski

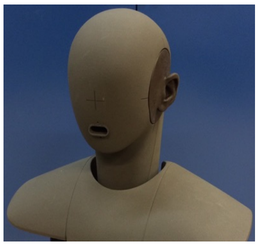
Outline

- Introduction
- Theory part
  - Adaptive filters, HP Target Curves, Augmented Reality Audio
- Prototype Headset
- Example 1: Adaptive Hear-Through Algorithm
- Example 2: Adaptive HP Equalization
- Conclusion

Introduction

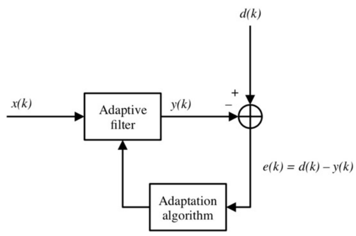


- About me:
  - M.Sc. from Aalto in 2016
  - Final-year doctoral student
  - Research topics: audio signal processing, headphones & loudspeaker equalization, equalizer design
  - Collaborations eg. with Nokia & Genelec



- Today's topics: augmented reality audio (ARA), headphones, equalization, adaptive filters

Adaptive Filters



- Time-varying filters
  - Coefficients are modified according to the input signal
- Two signals: input *x* & desired signal *d*
- Algorithm tries to minimize an objective function
  - E.g. mean-square error $E[|e(k)|^2]$
- Important factors: rate of convergence, misadjustment, tracking ability, robustness, computational requirements, numerical properties
- Four classes: system identification, inverse modeling, signal enhancement, and prediction

LMS Algorithm

- Introduced in 1960 by Widrow and Hoff
- One of the most popular adaptive filter algorithms
  - Computational simplicity and effectiveness
- Gradient-based algorithm

$$\mathbf{w}(k+1) = \mathbf{w}(k) + 2\mu e(k)\mathbf{x}(k)$$

- Coefficient vector can be initialized to zero or some estimate values
- Step size μ affects stability, convergence speed, and misadjustment

Extensions to LMS algorithm

- NLMS algorithm
  - Normalized Least Mean Square
  - In LMS, the compromise between speed and error due to the step size → time-varying step size
  - Introduced by both Nagumo and Noda and Albert and Gardner in 1967

$$\mu_{\text{NLMS}} = \frac{\mu}{\mathbf{x}^T(k)\mathbf{x}(k)+\epsilon}$$

  - Improved convergence speed
- Sign algorithm
  - If faster adaptation process is needed, only sign of error is used
  - Drawback: larger steady-state error

Extensions to LMS algorithm

$$\mathbf{w}(k) = \mathbf{w}(k-1) + \min\left[\frac{|e(k)|}{\|\mathbf{x}(k)\| + \epsilon}, \sqrt{\delta(k-1)}\right] \mathrm{sign}[e(k)] \frac{\mathbf{x}(k)}{\|\mathbf{x}(k)\| + \epsilon}$$

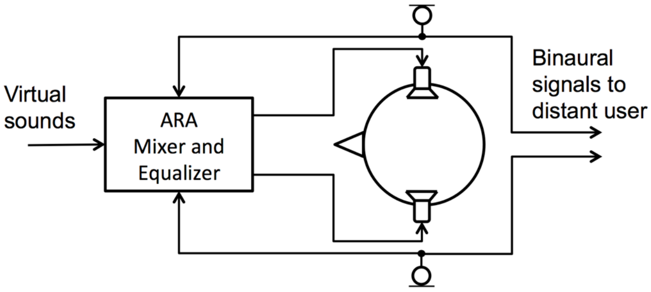
- RVSS-NLMS algorithm (Vega *et al.* 2008)
  - Robust: weakly sensitive to large perturbations in the input signal
    - *The filter coefficient update is limited at each iteration by a convergent positive sequence δ(k)*
  - Improved convergence rate when compared to NLMS
  - Changes operation between NLMS and NSA depending on the min-funtion OR can be viewed as NLMS with variable step size

$$\mu = \min\left[\frac{\sqrt{\delta(k-1)}}{|e(k)|/\|\mathbf{x}(k)\|}, 1\right]$$

Extensions to LMS algorithm

- Parameters of RVSS-NLMS:
  - δ sequence: initially large values, low values in the end
  - $\delta(0)$: the sequence is calculated recursively → initial value needed (affects the convergence rate)
  - Memory factor α: controls the tradeoff between convergence rate and robustness
  - Parameter κ: depends on the color of the input signal

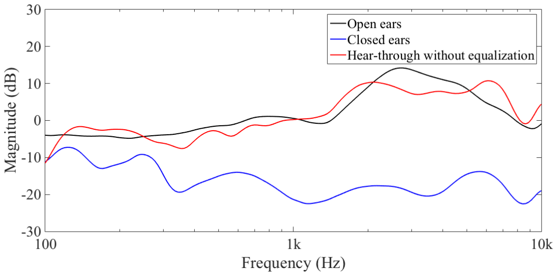
Augmented Reality Audio



Acoustical Transparency

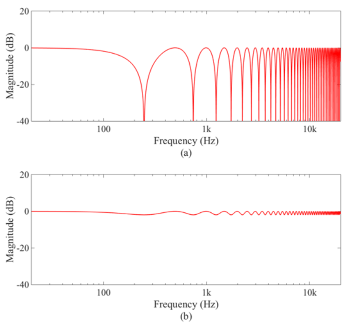
- Ideally, ambient sounds are perceived similarly when wearing headphone as with open ears
- However, headphones affect the perception of sounds
  - Attenuation:
  - In addition, in-ear headphones alter the acoustics of the outer ear
    - *Open ears: ear canal acts as quarter-wavelength resonator*
    - *Closed ears: ear canal acts as half-wavelength resonator*

Acoustical Transparency



- Therefore, equalization is needed
  - Performed by the ARA mixer

Comb-Filtering Effect

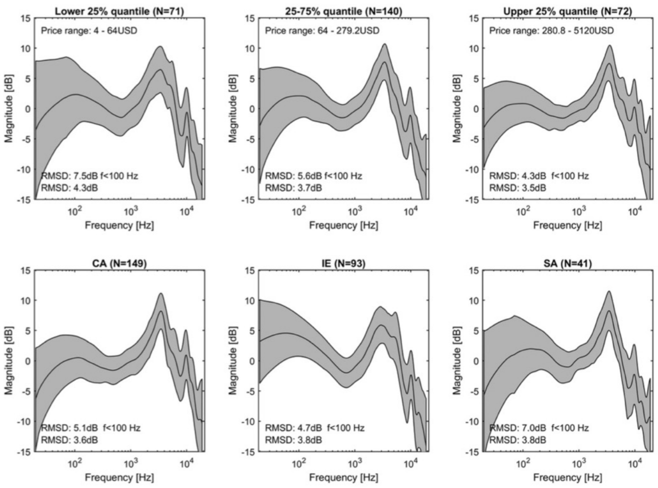


- Delayed version of sound is summed with the original
- The notches depend on the time delay and the magnitude of the signals
  - Delay affects the locations and magnitudes the size of the notches

➢ One of the signals must be attenuated

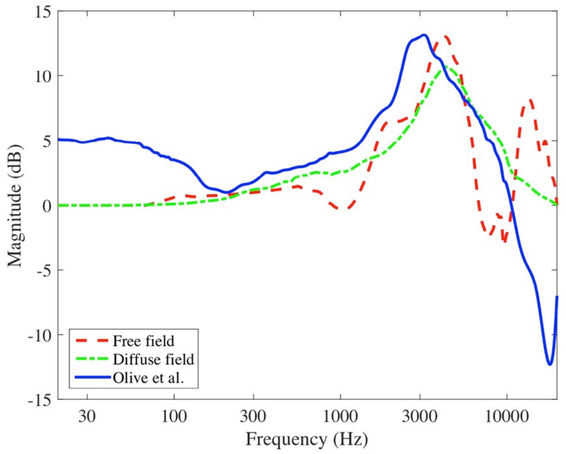
Headphone Frequency Responses

- Responses relative to price range (upper row) and type (lower row)



Ref: J. Breebaart, “No correlation between headphone frequency response and retail price,” 2017.

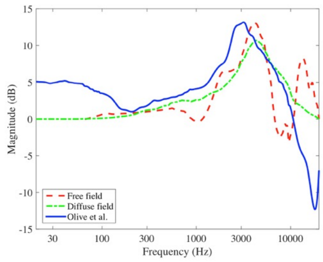
Headphone Target Curves



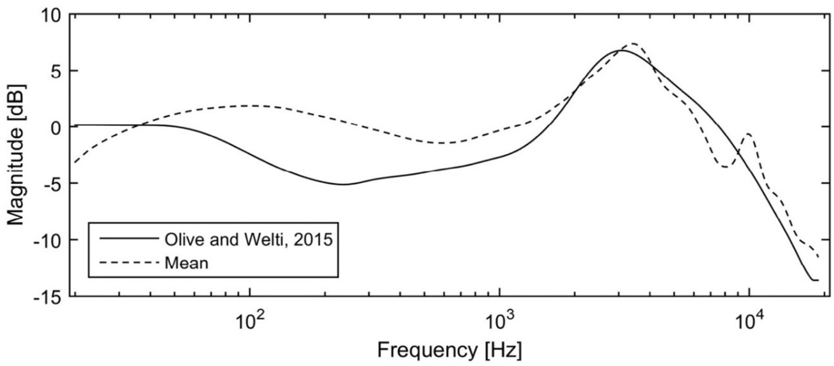
Ref: V. Välimäki & J D. Reiss, “All about audio equalization: Solutions and frontiers,” 2016.

Headphone Target Curves

- Unlike loudspeakers, there are many different ones
  - No accepted standard
- Free field
  - Sound source in an anechoic room
- Diffuse field
  - "An omnidirectional loudspeaker located far away"
- Recent work of Olive *et al.*
  - Listener preferred
  - A high-quality speaker in a room
  - Depends on the headphone type

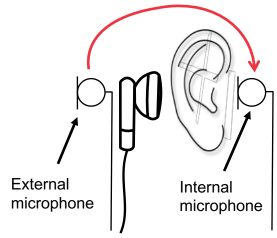


Headphone Target Curves

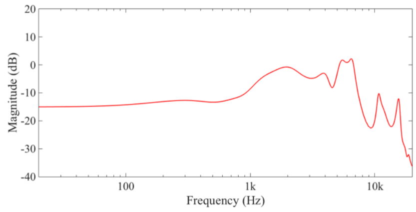


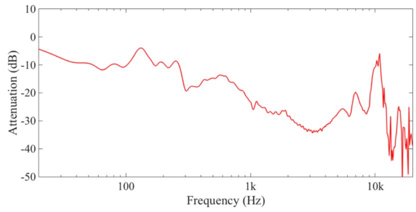
- However, single target may not be enough due to differences in headphone type, audio content, and listener demographics.

Prototype Headset



- Headset specifically designed for the projects
- Augmented reality audio headset with two microphones
  - Outside and inside the ear canal when the headset is worn
- In-ear headphones with balanced armature transducers





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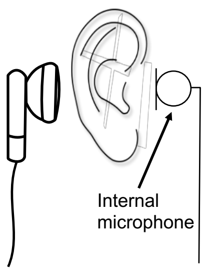
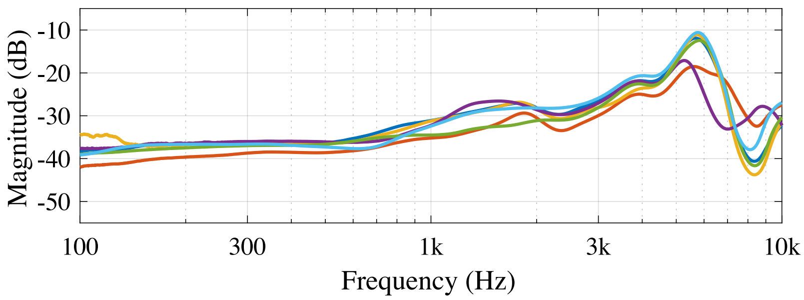
Prototype Headset



- Microphones inside ear canals can be used to measure individual HP response at the closed ear canal entrance
- Unique fit:

Frequency responses measured with headset microphone



Adaptive Hear-Through Algorithm

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Adaptive Hear-Through Algorithm

- Liski, J., Väänänen, R., Vesa, S., & Välimäki, V. (2016, August). Adaptive equalization of acoustic transparency in an augmented-reality headset. In AES Int. Conf. on Headphone Technology.

Audio Engineering Society
Conference Paper
Presented at the Conference on
Headphone Technology
2016 Aug 24–26, Aalborg, Denmark

Adaptive Equalization of Acoustic Transparency in an Augmented-Reality Headset

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ABSTRACT

Headphones are commonly used in noisy environments. Insert headphones attenuate and color the spectra of ambient sounds and thus alter the auditory perception. When the ambient sounds are desirable, a hear-through function can be used to reproduce them naturally while wearing headphones, i.e. to make the headphones acoustically transparent. A novel adaptive hear-through algorithm is proposed, which estimates the isolation and fine-tunes the hear-though equalization for optimal acoustic transparency. Measurements on a prototype headset and simulations show that the proposed algorithm produces acoustic transparency with default settings when the fit is good, and that the adaptation improves the transparency by up to 6 dB when the headset is poorly fitted. Volume control with additional shelving filter adjustments reduces the comb-filtering effect at frequencies below 1 kHz. The proposed algorithm is a suitable premise for augmented reality audio applications and offers improved behavior when compared to fixed hear-through systems.

1 Introduction

Headphones are nowadays used in various environments, resulting in an increased exposure to ambient sounds while wearing them. In addition to unwanted noises, ambient sounds can also be desirable and even vital; speech and siren are respective examples. Therefore, even though the attenuation properties of headphones offer health benefits [1], they may be a nuisance, since the headphones must be removed to hear the sound environment properly. A more convenient solution would eliminate the necessity to remove the headphones, and this is the idea behind augmented reality audio (ARA) and hear-through (HT) technology [2, 3, 4, 5, 6].

In augmented reality audio, real sound environments are extended with virtual auditory environments [2]. ARA applications are based on the idea of constantly wearing headphones. The premise for this is a HT function, which plays a processed version of the ambient noise through the headphone to simulate hearing with open ears [2], i.e. it makes the headphones acoustically transparent. The HT function can be used to better integrate headphones into our everyday life, since otherwise headphones alter the acoustics of the outer ear [7], which results in an altered listening experience. The HT function can also be used to hear ambient sounds selectively while listening to music [8, 9] and for improving the sound quality in live concerts [10].

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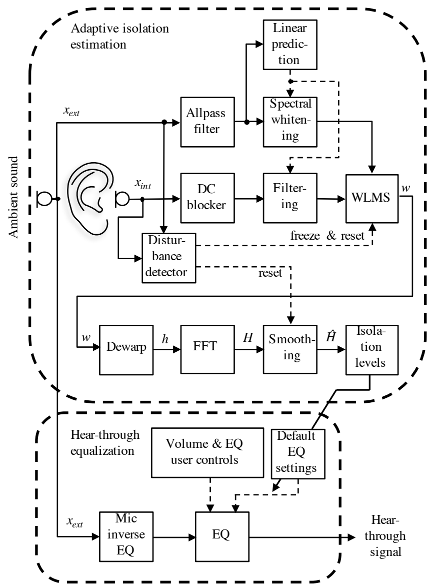
Goals

- Measure the characteristics of the headset
- Develop an algorithm to estimate isolation between internal and external microphone
- Define the required equalization to transform HP acoustically transparent
- Use the estimate to adapt hear-through equalization
- Enable user controls for EQ and volume
  - Take comb-filtering effect into account
- Verify functions with measurements

Proposed Algorithm

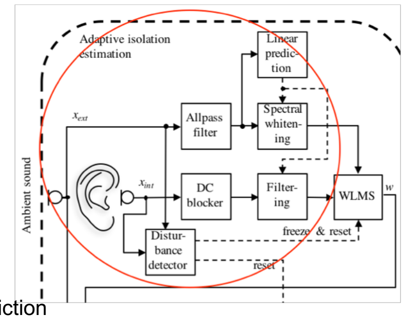
- Two parts:
  - Adaptive Isolation Estimation
  - Hear-Through Equalization
- **Both the estimation and the equalization are adaptive**
- Interesting frequency band is limited to 100-10000 Hz



Adaptive Isolation Estimation

Preprocessing:

- DC Blocker
  - Removes heartbeat and other noise
  - In other signal, allpass filter is used to compensate for the group delay of the dc blocker
- Spectral Whitening
  - All-zero inverse filter *A(z)* with linear prediction
  - To improve the speed of NLMS algorithm



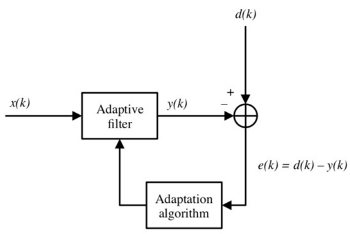
Disturbance Detection

- Freezes the algorithm and resets certain variables

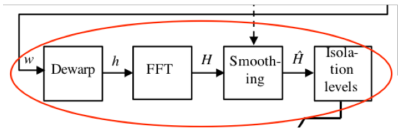
Adaptive Isolation Estimation

- The isolation estimation with warped RVSS-NLMS algorithm (robust variable step-size normalized least mean square)
- External microphone signal is input signal *x(k)*
- Internal microphone signal is desired signal *d(k)*

➢ System identification: output of the adaptive filter models the behavior of the earpiece isolation

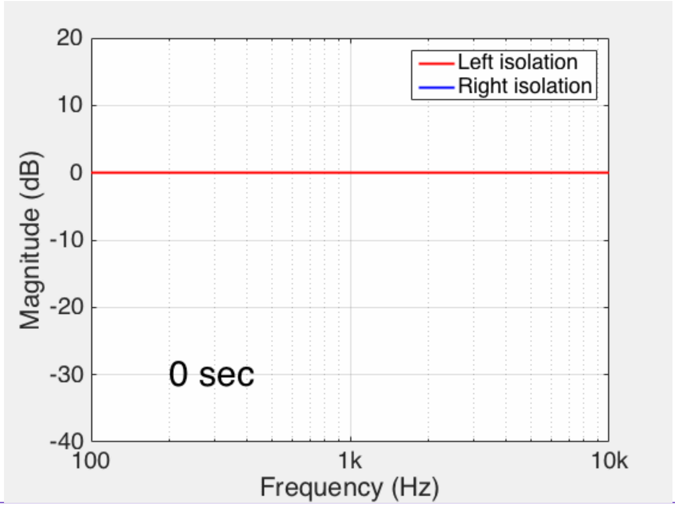


Adaptive Isolation Estimation

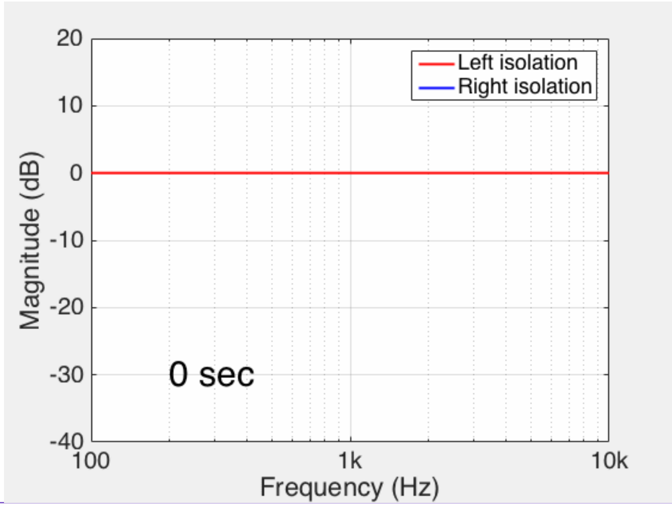


- Dewarping
  - Transform back to linear frequency scale
- Smoothing
  - 1/3 octave smoothing and averaging
- The output of the first part: **Isolation Levels**
  - The estimated isolation in dB
  - Fed to the second part of the algorithm for equalization control

Adaptive Isolation Estimation

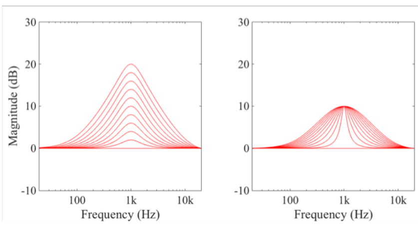


Adaptive Isolation Estimation



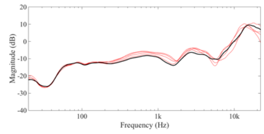
Hear-Through Equalization

- All equalization is implemented with digital Regalia-Mitra parametric equalizers

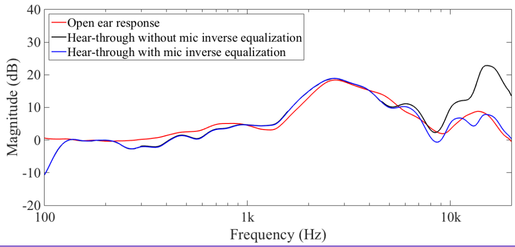


- 1st-order shelving filters
  - Gain & Cutoff frequency
- 2nd-order peaking filters
  - Gain, Cutoff frequency & Bandwidth

Hear-Through Equalization

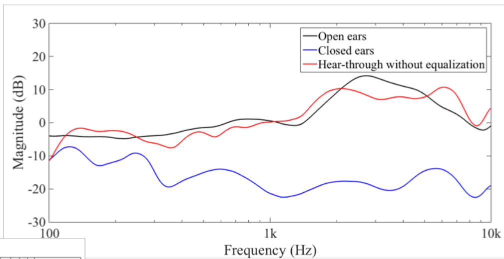


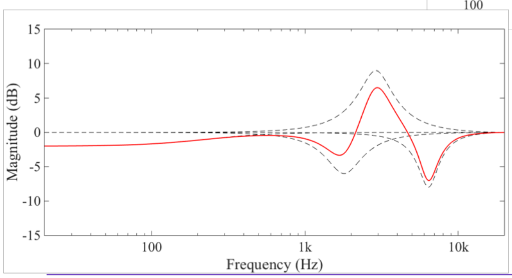
- Microphone inverse processing
  - Since the microphones are attached to circuit boards, boost occurs between 10-20 kHz ➔ tinny HT-signal
  - Equalized with second-order Regalia-Mitra filter with constant parameters:



Hear-Through Equalization

- Default target derived from free-field measurements
  - Frontal sector, 0–30˚

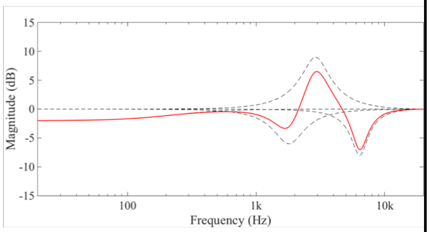




- 4 Regalia-Mitra equalizer sections

Hear-Through Equalization

- Actual Hear-through equalization
  - Default settings derived from measurements to produce acoustical transparency when HP are properly fitted
    - 1 shelf, 3 peak/notch filters



- Adaptation according to the Isolation Levels
- User controls for EQ
- User control for HT signal volume

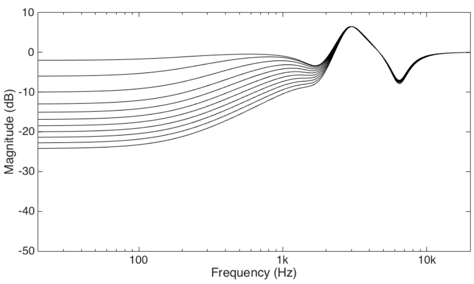
➢ Comb-filtering effect must be taken into account with adaptation and volume control

Original sound file

Leaked sound

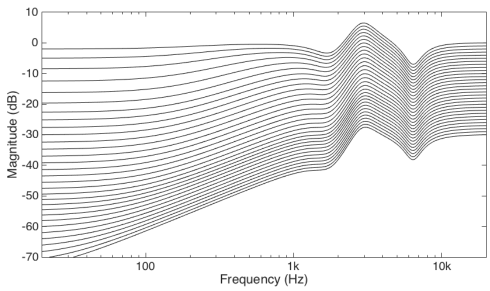
Leaked sound + HT signal

Equalization Controls

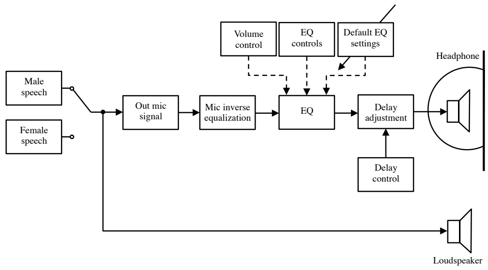


- Automatic adaptation

- Volume control

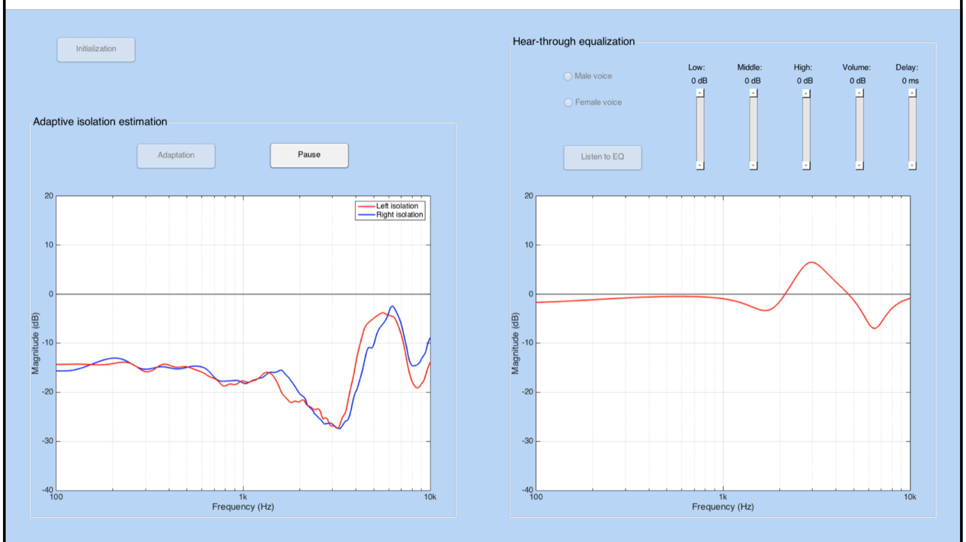


Matlab Simulator



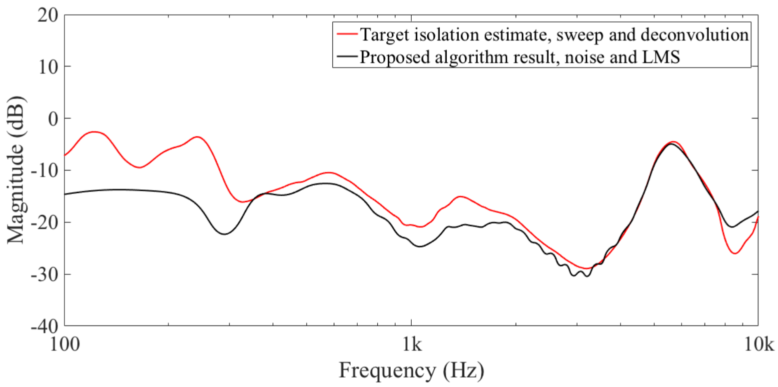
- Realization differs slightly from the proposed block diagram
  - Matlab doesn't allow real-time implementation ➔ Simulation
  - Delay can be additionally controlled
- Pinkish traffic-like noise to estimate isolation and speech signal to demonstrate the hear-through part of the algorithm

Matlab Simulator



Measurement Results

Results: Isolation estimate



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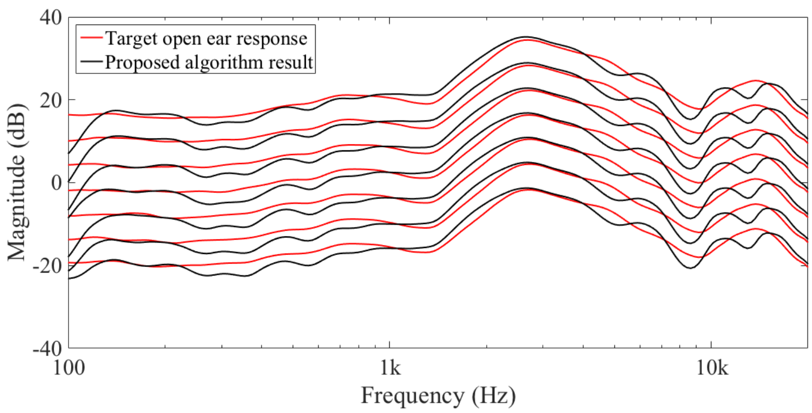
Results: Hear-through equalization



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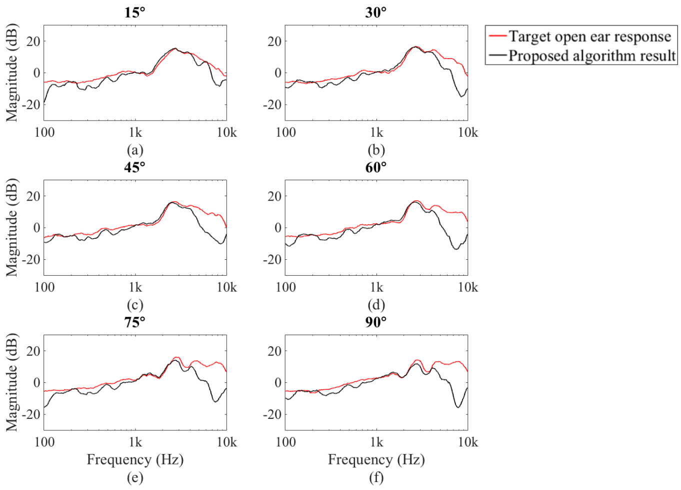
Results: Other directions



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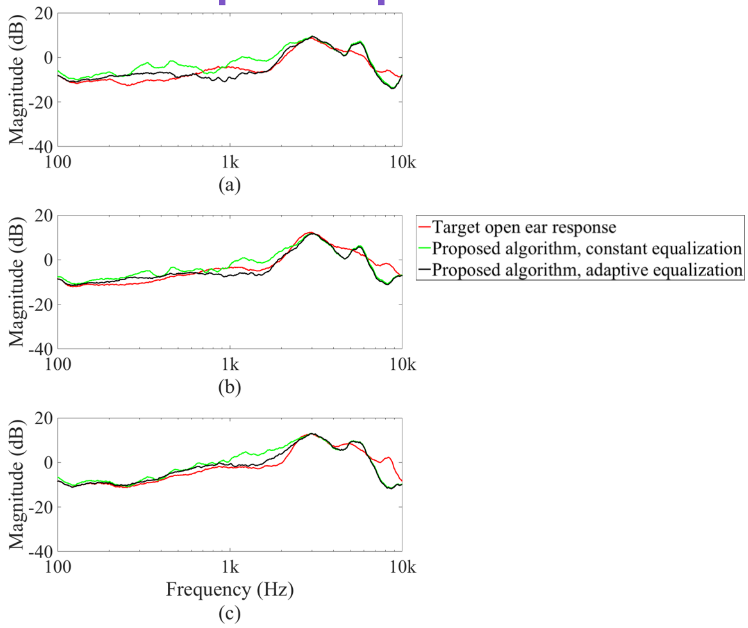
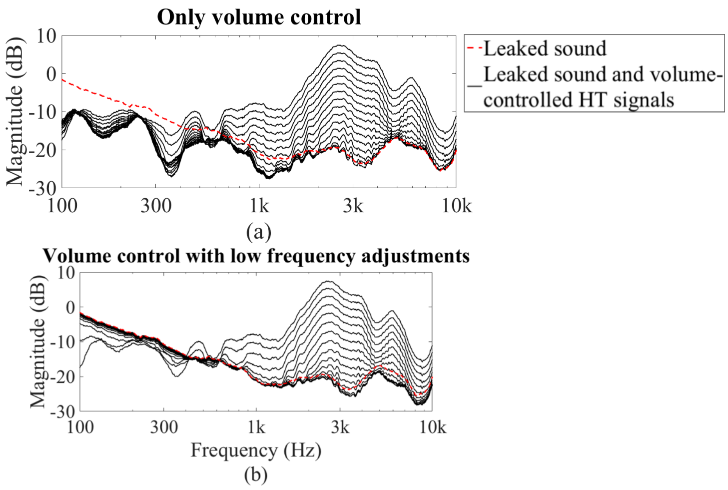
Results: Adaptive equalization



Results: HT signal volume control



Adaptive Headphone Equalization

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Adaptive HP Equalization

- Liski, J., Välimäki, V., Vesa, S., & Väänänen, R. (2017, August). Real-time adaptive equalization for headphone listening. In *25th European Signal Processing Conference (EUSIPCO)*.

Real-Time Adaptive Equalization for Headphone Listening

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Abstract—**The experienced sound quality produced by headphones varies between individuals. Especially with insert headphones a constant equalization may not work properly, when a specific perceived frequency response is desired. Instead, adaptive individualized equalization can be used. Previously this required multiple sensors in a headphone earpiece. This paper proposes a signal processing algorithm for continuous on-line equalization of a headset with a single microphone. The magnitude response of the headphones is estimated using arbitrary reproduced sounds. Then, the headphone response is equalized to a user-selected target response with a graphical equalizer. Measurements show that the proposed algorithm produces accurate estimates with different sound materials and the equalization produces results that closely match the target response. The algorithm can be implemented for multiple applications to obtain accurate and quick personalization, since the target response can be set arbitrarily.**

I. Introduction

Headphones are now widely used to listen to reproduced sounds such as music or binaural sounds. In music listening, the source material is usually intended for stereo loudspeaker playback, and in binaural reproduction, the precise spectral balance at the eardrum of the listener is paramount in order to produce the desired perception. Thus, in both applications the frequency response of the headphones is important due to its effects on the listening experience. Furthermore, since there is no industry standard for the target frequency response, the headphones may need to be equalized to match the required frequency response needed for each application.

One popular headphone type is insert headphones that are placed partly inside the ear canal. They couple directly to the ear canal and produce a unique fit for each person every time they are inserted into the ear, since the ear canals vary between individuals. This varying fit results in varying experienced sound quality even when using the same pair of headphones. Therefore, a similar equalization may not work for everybody, but instead a personalized equalizer (EQ) is required. To design a personal EQ curve, one needs to know the sound pressure at the eardrum.

One way to estimate the sound pressure at the eardrum relies on solving the ear canal parameters, such as the impedance of the ear canal or the eardrum [1], [2], [3]. The parameters are used to construct a physics-based computational model of the ear canal [4], which in turn enables the pressure at the eardrum to be estimated based on the pressure at the ear canal entrance. The downside is the arduous impedance measurement, which is not suitable for everyday use. Another method utilizes measurements of both the sound pressure and the velocity. Hiipakka *et al.* proposed an accurate method, which estimates the pressure at the eardrum from measurements performed at the ear canal entrance [5], [6]. The method, however, requires an extra-small sound-velocity probe in addition to the normal sound pressure probe, which thus increases the complexity and cost of the headset. State-of-the-art adaptive headphones calibrate the frequency response only once [7].

This work presents an adaptive equalization system for headphone listening. The system includes a prototype headset, which contains microphones inside the ear canal when the headset is worn. These internal microphones are used during the sound reproduction, and the resulting recording is utilized to estimate the frequency response of each earpiece at the microphone location. The estimate is then mapped to the eardrum with an ear canal model. Finally, a third-octave graphic EQ is designed according to the user-defined target response to obtain a personalized frequency response.

There are many applications where individualized equalization is desired. In addition to music listening and binaural reproduction, specific equalization is required if one pair of headphones is used to simulate another pair. Rämö *et al.* proposed a method to simulate different headphones in a noisy environment, when the frequency response and the isolation properties of the simulated headphones are known [8]. Thus, new headphones could be tested before purchase in order to test their suitability for outdoor listening. Similarly, Olive *et al.* proposed a method to perform double-blind headphone listening tests with a single pair of headphones [9], which enables better comparison of sound quality, since the aesthetics and the feel of the headphones do not affect the listening.

The structure of the paper is as follows. The proposed algorithm is introduced and its building blocks are analyzed in Section II. Section III presents measurement results to illustrate the behavior of the proposed algorithm. Finally, Section IV concludes this paper.

II. Adaptive-Equalization Algorithm

This section presents a novel signal processing algorithm to estimate the individual frequency responses of a pair of

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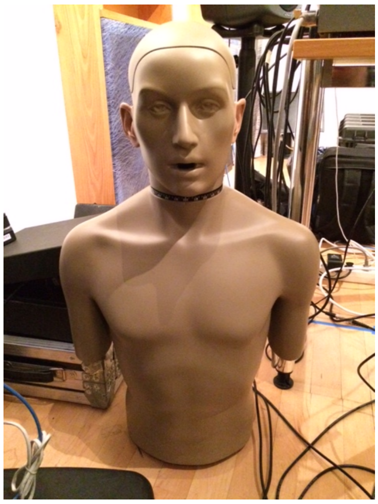
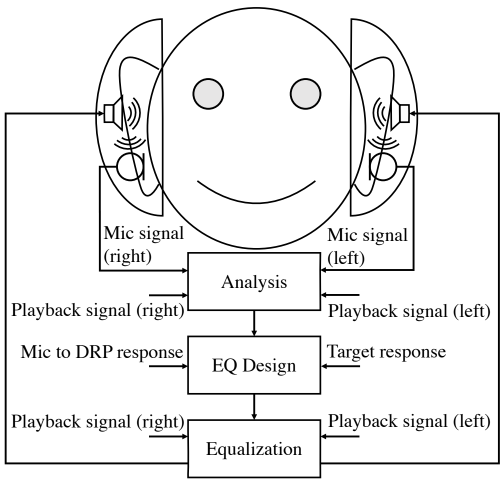
Motivation

- No industry standard for the headphone target frequency response
- Equalization can be used to obtain a desired frequency response

➢ A novel adaptive equalization algorithm for headphone listening



Proposed Algorithm

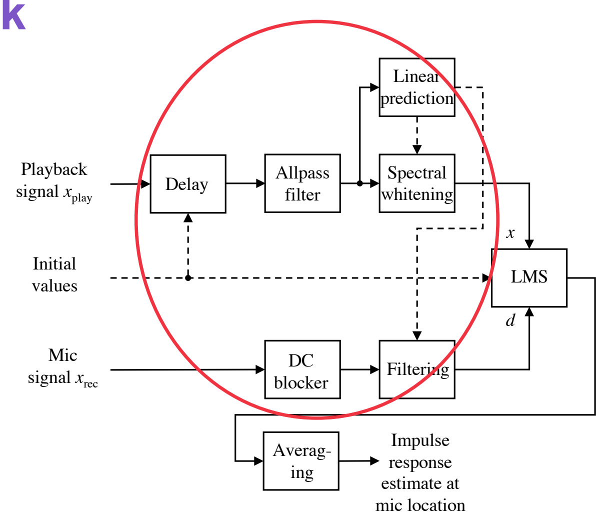


Frequency range:
100 Hz – 10 kHz

Analysis Block

Preprocessing

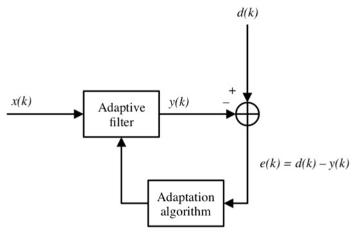
- DC Blocker
  - Removes noise
- Spectral Whitening
  - Improves estimation speed



Analysis Block: Estimating the frequency response

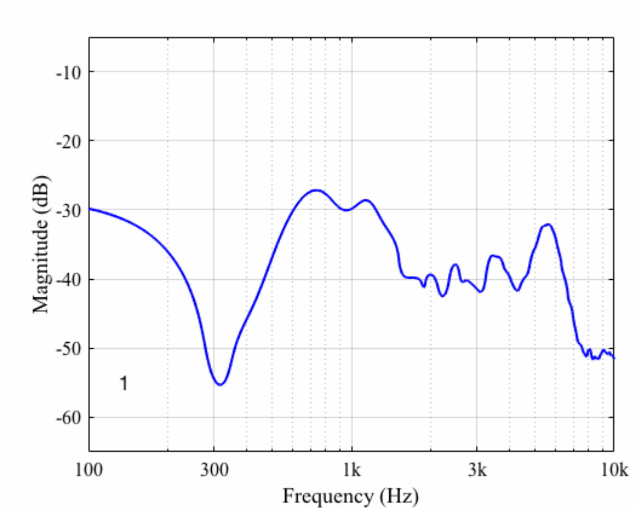
$$\mathbf{w}(k) = \mathbf{w}(k-1) + \min\left[\frac{|e(k)|}{\|\mathbf{x}(k)\| + \epsilon}, \sqrt{\delta(k-1)}\right] \mathrm{sign}[e(k)] \frac{\mathbf{x}(k)}{\|\mathbf{x}(k)\| + \epsilon}$$

- Once again, RVSS-NLMS algorithm is used
- Input *x* is the playback signa
- Desired signal *d* is the internal microphone signal
- After the convergence, the filter coefficient correspond to the impulse response at the mic location



FR Estimation

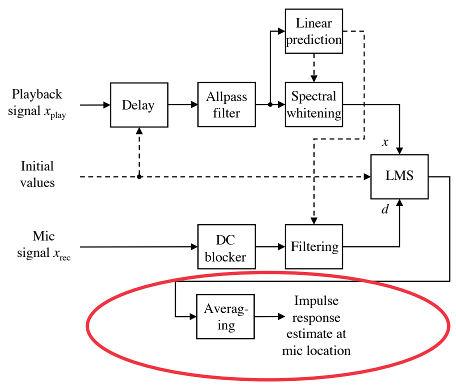
- Using music as adaptation signal
- Adaptation in blocks
- No averaging → large deviations especially at low frequencies



Analysis Block

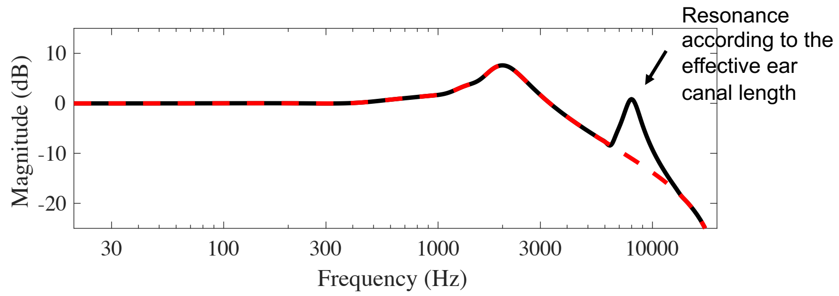
Output of the Analysis block

- Estimated headset IR
- Back to LMS block
- Forward to EQ Design block



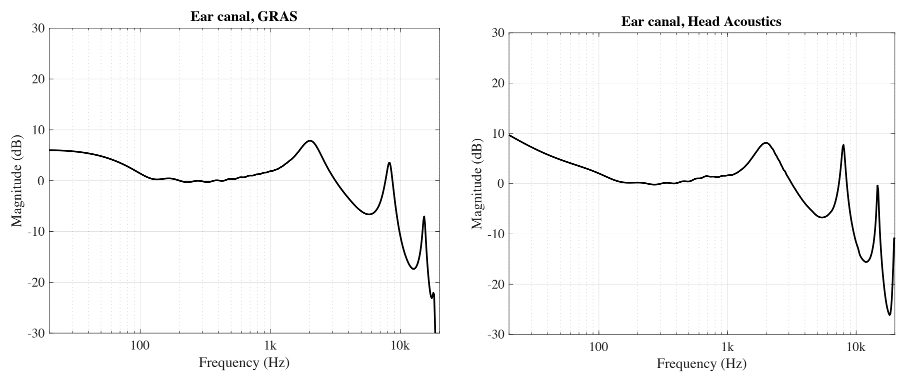
EQ Design Block

- Maps the estimate from mic locations to eardrum
- Obtained from two dummy heads



Red: Default shape
Black: Example response, ear canal length 22 mm

EQ Design Block

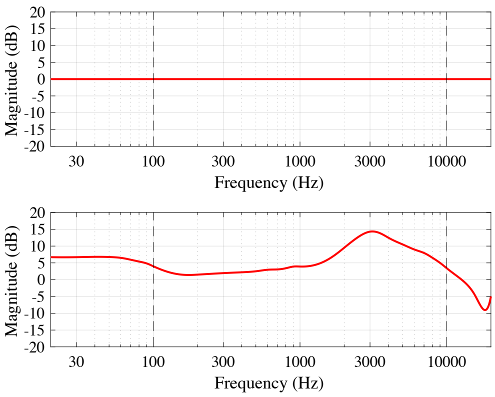


Other methods for Ear Drum Response

- Measure the individual headphone transfer function (HpTF) off-line at the eardrum with probe microphones
- Estimate the pressure at the eardrum from sound pressure and velocity measurements performed at the ear canal entrance
- Model the ear canal based on the impedance of the ear canal and the eardrum
- Subjective loudness matching using a reference signal

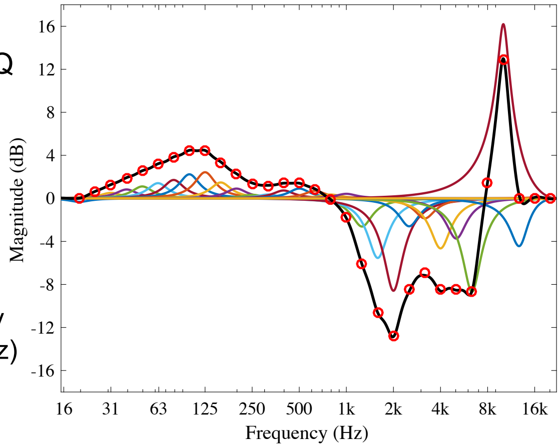
EQ Design Block

- Target responses
  - Flat response
  - Listener-preferred target (Olive *et al.*, 2013)
- Can be selected arbitrarily

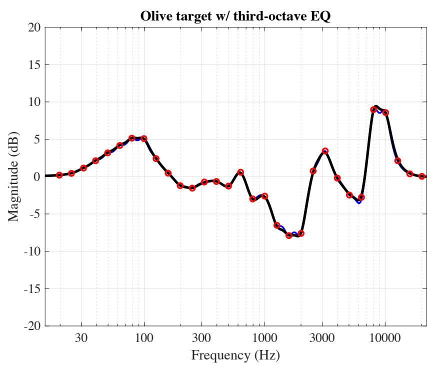


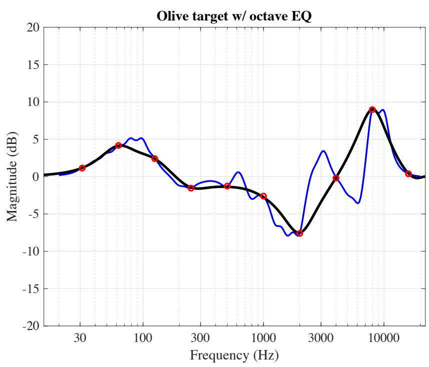
EQ Design Block

- Third octave graphic EQ proposed by Liski & Välimäki (2017)
  - Accurate & simple
- 8 lowest band & 3 uppermost band tweaked
  - Due to used frequency band (100 Hz – 10 kHz)

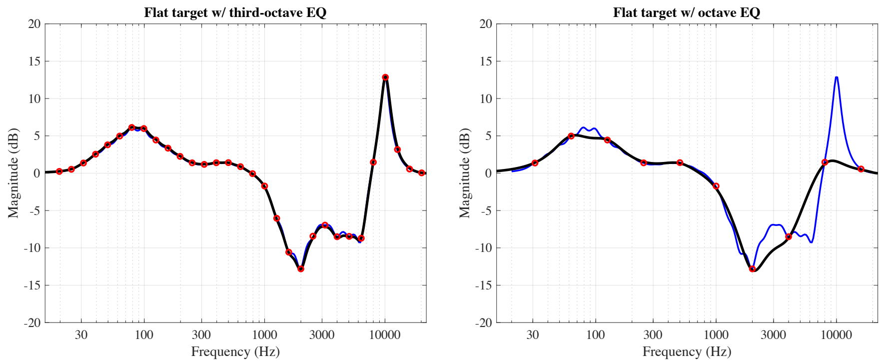


Octave EQ vs Third-octave EQ

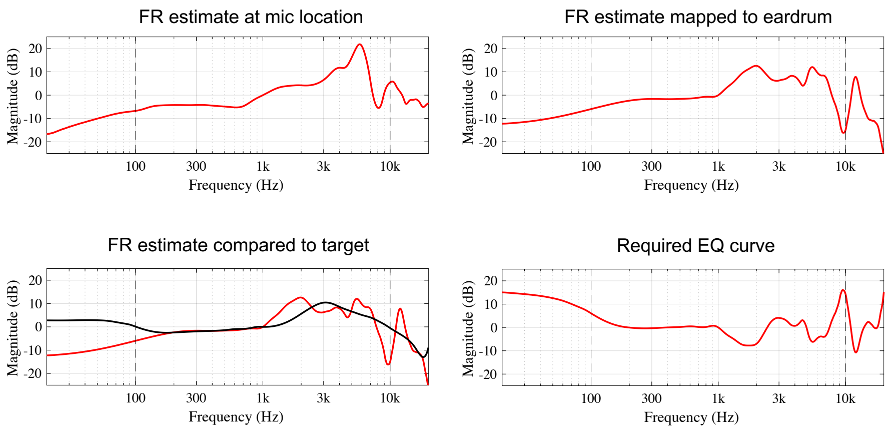




Octave EQ vs Third-octave EQ

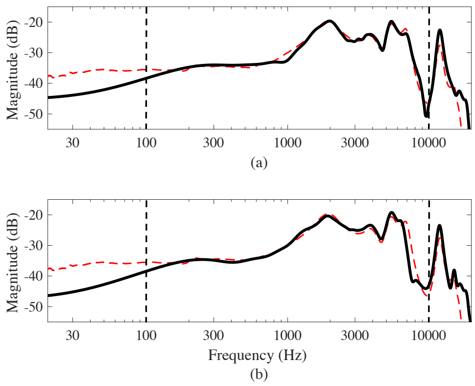


Analysis Steps



Results

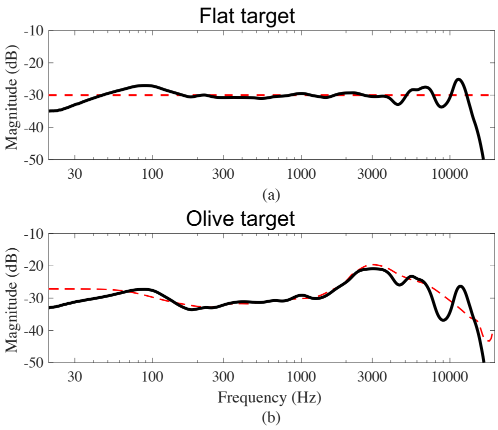
- Frequency response estimation accuracy
- Two different music signals used
  - Accurate with multiple types of signals



Red: Sweep measurement (target)
Black: Algorithm estimate

Results

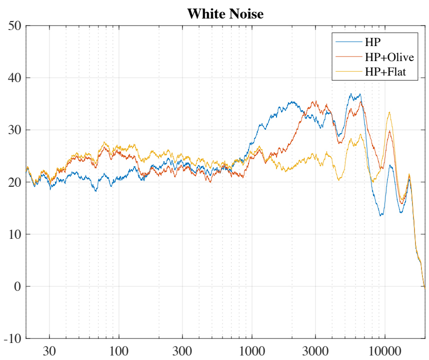
- Equalization accuracy
- Measured with sweep using a dummy head
- Algorithm provides the desired effect

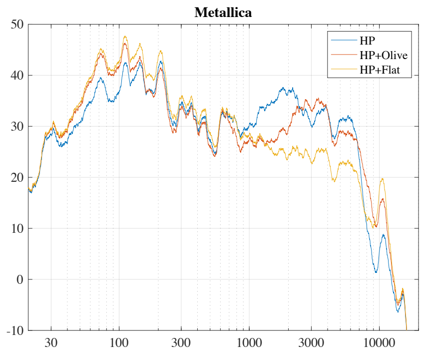


Sound Examples

	Through HP	HP+OliveEQ	HP+FlatEQ
Metallica			
Norah Jones			
White noise			
Pink noise			

Sound Examples





Conclusion

- Prototype ARA headset
  - Algorithms specific for insert headphones
- Adaptive filters were used to estimate headphone isolation and frequency response
- Working with arbitrary signals
  - Isolation with ambient noise
  - Response with music
- Headphone equalization is a powerful tool to change the sound or enable useful applications