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A ZERO-PHASE PARAMETRIC EQUALIZATION FOR MICROPHONE AND LOUDSPEAKER IMPULSE RESPONSE

Victor Popa[1*] **Bogdan Moroșanu**[1] **Horia Sebastian Ioniță**[1]
[1] Department of Telecommunications, University Politehnica of Bucharest, Romania

ABSTRACT

When measuring acoustic room impulse responses a known problem is the elimination of the influence of the loudspeaker and microphone. A solution is to estimate an inverse filter that flattens the spectrum of the transfer function. This paper proposes a method for equalizing the spectrum of the measuring system with a zero-phase parametric filter in order to eliminate the influence of the microphone and loudspeaker. At first the measured impulse response is sectioned to extract the direct sound which corresponds to the microphone and loudspeaker response. The parameters for the inverse zero-phase filter are obtained with a two stage optimization process: the first one matches the spectral envelope and the second one matches the octave band energy so that the resulting spectrum does not alter the perceived sound in spatialization applications. Tests are done to prove the efficiency of the filtration with both room impulse responses and head-related impulse responses.

Keywords: *zero-phase filtering, parametric equalizer, impulse response correction.*

1. INTRODUCTION

Acoustic impulse responses are essential for applications in architectural acoustics, spatial audio, and virtual reality [1]. A persistent challenge in obtaining accurate measurements is the influence of non-ideal frequency responses from microphones and loudspeakers, which inevitably color the measured responses [2]. These spectral artifacts can significantly impact applications requiring perceptual accuracy, such as auralization systems [3].

The frequency response characteristics of transducers introduce spectral artifacts that do not belong to the acoustic space under investigation [4]. While sophisticated measurement equipment can mitigate some of these effects, the complete elimination of transducer influence remains a challenging problem in acoustics. In room acoustics and spatial audio applications, preserving the temporal structure of impulse responses is particularly crucial. Parameters like initial time delay gap, reflection patterns, and reverberation time directly influence spatial perception and require phase preservation for accurate analysis [5].

Zero-phase equalization maintains these temporal relationships intact while still compensating for spectral colorations, effectively isolating the acoustic properties of the measured space from the measurement system's transfer function. State-of-the-art techniques include FIR-based linear-phase equalizers [6], which inherently preserve phase linearity while enabling precise amplitude equalization. Contemporary studio monitors integrate advanced DSP algorithms [7] that utilize both FIR and IIR filters to simultaneously address amplitude and phase distortions. Kautz filters represent another sophisticated approach tailored for audio equalization, supporting direct least squares optimization for high-precision equalization of loudspeaker and room responses.

However, several challenges remain. The efficacy of equalization filters depends heavily on the precision of impulse response measurements, with inaccurate or noisy measurements potentially leading to erroneous filter designs. Zero-phase equalization, particularly with long-tap FIR filters, imposes significant computational demands. Additionally, the dynamic nature of acoustic en-

*__Corresponding author__: victor.popa@upb.ro.

vironments presents challenges for static equalization filters, as changes in room geometry or listener position can alter acoustic responses [7].

This paper presents a method for equalizing the spectrum of the measuring system using zero-phase parametric filtering to eliminate transducer influence while preserving temporal characteristics. Our approach uses a two-stage optimization process: first matching the spectral envelope, then matching octave band energy to maintain perceptual fidelity in spatial applications.

The paper is structured as follows. Section 2 formulates the mathematical framework for impulse response measurement and details our methodology for direct sound extraction and parametric filter design. Section 3 presents the zero-phase implementation approach, optimization process, and algorithm design. Section 4 demonstrates experimental results from room impulse responses and head-related transfer functions. Section 5 offers conclusions and contextualizes our findings within acoustic measurement techniques.

2. PROBLEM FORMULATION

Let us consider a measurement system for acquiring acoustic impulse responses, consisting of a loudspeaker, a microphone, and the acoustic environment under test. The measured discrete impulse response $h_{\text{meas}}(n)$ can be expressed as the convolution of multiple subsystem responses:

$$h_{\text{meas}}(n) = h_{\text{ls}}(n) * h_{\text{room}}(n) * h_{\text{mic}}(n), \tag{1}$$

where $h_{\text{ls}}(n)$ represents the loudspeaker impulse response, $h_{\text{room}}(n)$ the actual room impulse response, and $h_{\text{mic}}(n)$ the microphone impulse response [4]. In the frequency domain, this relationship becomes multiplicative:

$$H_{\text{meas}}\left(e^{j\omega}\right) = H_{\text{ls}}\left(e^{j\omega}\right) \cdot H_{\text{room}}\left(e^{j\omega}\right) \cdot H_{\text{mic}}\left(e^{j\omega}\right). \tag{2}$$

The objective of measurement system equalization is to extract $H_{\text{room}}\left(e^{j\omega}\right)$ by compensating for the transducer responses:

$$H_{\text{room}}\left(e^{j\omega}\right) = \frac{H_{\text{meas}}\left(e^{j\omega}\right)}{H_{\text{ls}}\left(e^{j\omega}\right) \cdot H_{\text{mic}}\left(e^{j\omega}\right)}. \tag{3}$$

However, direct inversion presents several challenges. The combined transducer response given by $H_{\text{trans}}\left(e^{j\omega}\right) = H_{\text{ls}}\left(e^{j\omega}\right) \cdot H_{\text{mic}}\left(e^{j\omega}\right)$ typically exhibits spectral nulls at certain frequencies, making naive inversion numerically unstable [8].

The temporal structure of $h_{\text{room}}(n)$ is critical for acoustic characterization, containing essential information about reflections, reverberation decay, and spatial cues [1]. Conventional equalization filters with non-linear phase responses distort this temporal structure, compromising both spatial localization and derived acoustic parameters such as reverberation time (T_{60}) and clarity index (C_{50}) [5].

For head-related transfer functions (HRTFs), the situation is particularly demanding as interaural time differences and spectral cues are the primary localization mechanisms, requiring phase-preserving equalization [9].

We therefore establish the following criteria for effective measurement system equalization:

- The equalization filter must flatten the magnitude response within $(f_{\text{min}}, f_{\text{max}})$.
- The filter must preserve the temporal structure, implying a zero-phase characteristic.
- The equalization must be robust against measurement noise and numerical instabilities.
- The perceptual characteristics should remain consistent with the actual acoustic environment.

To address these requirements, we propose a zero-phase parametric equalization approach that isolates the direct sound component and applies a two-stage optimization process: first matching the spectral envelope, then preserving octave-band energy distribution. This approach avoids instabilities at spectral nulls while maintaining temporal relationships crucial for spatial audio applications.

3. FILTER ESTIMATION ALGORITHM

The measurement of the impulse response in obtained by using the exponential sweep method [2]. The first step is to create a zero-phase parametric equalizer with the transfer function $H_{\text{eq}}\left(e^{j\omega}\right)$ that equalizes the spectrum of the measurement system between angular frequencies ω_{min} and ω_{max}:

$$H_{\text{meas}}\left(e^{j\omega}\right) \cdot H_{\text{eq}}\left(e^{j\omega}\right) \approx H_{\text{room}}\left(e^{j\omega}\right), \tag{4}$$

with $\omega \in (\omega_{\text{min}}, \omega_{\text{max}})$.

To obtain the zero-phase filter we use the cascaded forward-backward filtering of the measured impulse response as presented in Fig. 1, where $H_{\text{peq}}(z)$ is the parametric equalizer transfer function, $h_f(n)$ is the forward filtered signals, $h_b(n)$ is the backward filtered signal and N is the length of the impulse response. The two "Reverse" blocks are used for temporal reversal of the samples. After a series of calculations we obtain the frequency response of the output of the filtering process as:

$$H_{\text{room}}\left(e^{j\omega}\right) \approx H_{\text{meas}}\left(e^{j\omega}\right) \cdot \left|H_{\text{peq}}\left(e^{j\omega}\right)\right|^2. \quad (5)$$

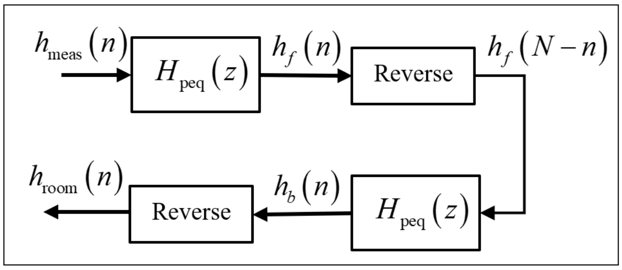

Figure 1. Cascaded zero-phase parametric equalizer block schematic.

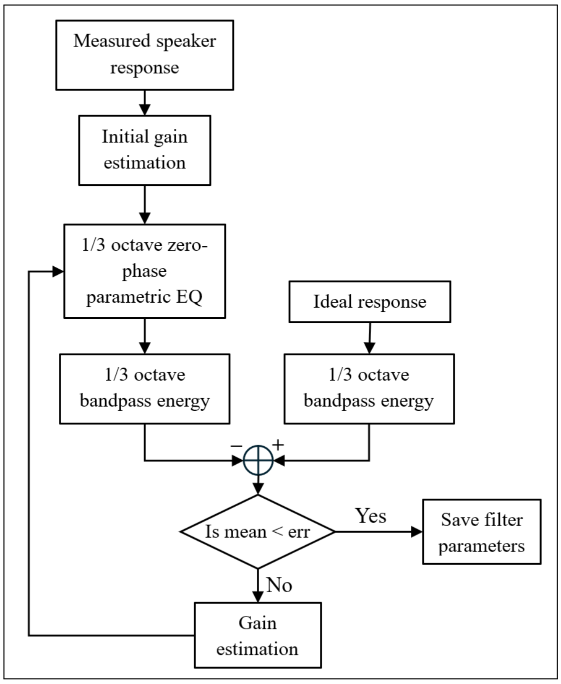

Figure 2. Speaker equalization algorithm.

The equalization of the measurement system is done in two steps. At first we equalize the speaker response with the algorithm described in Fig. 2. Since we want to separate it from the microphone response, we measured the speaker response in isolation with a calibrated measurement microphone that has a flat spectrum. The measured speaker spectrum is then $1/3$ octave smoothed and the initial gains for the parametric EQ are determined at the central frequencies. An $1/3$ octave bandpass filter is then applied in order to extract the energy on each subband and the results are compared to the ideal response bandpass energy.

All gain values are determined in dB and if the mean absolute gain for all the subbands is greater than a predefined threshold, err, the new gains are estimated and the process is repeated. We implemented the equalization by comparing the $1/3$ octave energy for two reasons: the octave energy better describes the perception of the human hearing apparatus; in spatialization applications the impulse response of the system should not affect the energy in each band.

The second equalization is done on the microphones used for measuring the HRTFs. The basic algorithm is very similar to the one in Fig. 2, where instead of the speaker response we use the ideal response and instead of the ideal response we use the gains defined in the microphone datasheet. After we determine the parameters for both the speaker and microphone equalization filters we cascade the two filters and apply them to measured head related impulse responses.

4. RESULTS

The measurement system we used is similar to the one described in [10]. The speaker used was a M-Audio BX8a studio monitor and the recording was done with binaural recording microphones from Brüel&Kjær type 4101 connected to a type 2250 Sound Level Meter that records at sample rate 48 kHz. Since the speaker response is between 150 Hz and 20 kHz we chose the minimum and maximum frequencies for the parametric equalizer and the octave bandpass filters as $f_{min} = 160$ Hz and $f_{max} = 16$ kHz. The parametric equalized is composed of second order, 1/3 octave bandwidth peak filters connected in cascade. The equalization of the speaker was done so that the mean absolute difference between the ideal response and the equalized response is smaller than 0.1 dB. We used the mean absolute value because for some frequencies the speaker response has a large attenuation, therefore we would not want to force equalization on frequencies that the speaker cannot reproduce.

The equalization of the speaker is presented in Fig. 3, where we notice that the filtered response with the determined parametric equalizer is flat in the frequency interval of interest.

The algorithm is then applied to the measured head related impulse responses and the equalization is done according to the datasheet of the microphones.

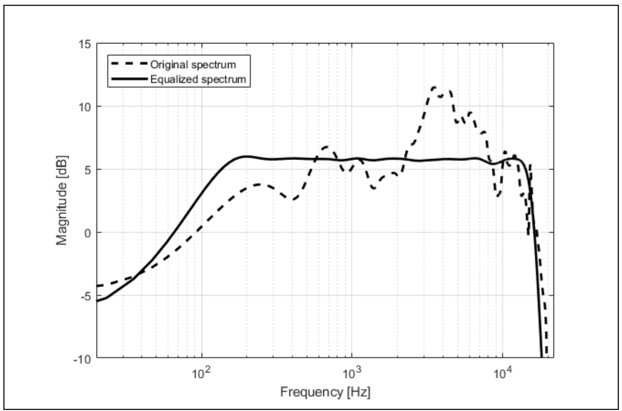

Figure 3. The result of the equalization. Dotted line - the original speaker spectrum. Solid line - the transfer function after equalization.

To further test if the zero-phase filtering is effective we calculated the difference between the original impulse response and the equalized impulse response phases and obtained a maximum difference of 0.024 rad, which is less that $1.5°$. The oscillations in phase difference can be attributed to numeric errors from calculating the phase, and we can conclude that the phase shift for the proposed filter is close to zero.

The same results were obtained after applying the algoritm to a room impulse response. The measured transfer function was first 1/3 octave smoothed and applied to the algorithm. The phase difference obtained after filtering was of 0.045 rad.

5. CONCLUSIONS

This article presented a zero-phase implementation of a parametric equalizer used to reduce the influence of loudspeakers and microphones in impulse response measurements that can be used in spatialization applications. The purpose of the filter is to minimize the alteration of the phase in room impulse response measurements and head related impulse responses when using imperfect equipment. The results suggest that the filter can be successfully applied both in HRTF measurements and room equalization applications. Further development is needed to test the equalized impulse responses in subjective tests to confirm their quality in sound localization applications.

6. REFERENCES

[1] H. Kuttruff, "Room acoustics," *CRC Press*, 2016.

[2] A. Farina *et al.*, "Simultaneous measurement of impulse response and distortion with a swept-sine technique," *Preprints-Audio Engineering Society*, 2000.

[3] D. Hammershøi and H. Møller, "Binaural technique—basic methods for recording, synthesis, and reproduction," *Communication acoustics*, pp. 223–254, 2005.

[4] M. Vorländer, "Acoustic measurements," in *Handbook of Engineering Acoustics*, pp. 23–52, Springer, 2012.

[5] J. Blauert, *Spatial hearing: the psychophysics of human sound localization*. MIT press, 1997.

[6] V. Välimäki and J. Reiss, "All about audio equalization: Solutions and frontiers," *Applied Sciences*, vol. 6, p. 129, 05 2016.

[7] K. Matti and T. Paatero, "Equalization of loudspeaker and room responses using kautz filters: Direct least squares design," *EURASIP Journal on Advances in Signal Processing*, vol. 2007, 01 2007.

[8] O. Kirkeby and P. A. Nelson, "Digital filter design for inversion problems in sound reproduction," *Journal of the Audio Engineering Society*, vol. 47, no. 7/8, pp. 583–595, 1999.

[9] P. Majdak, P. Balazs, and B. Laback, "Multiple exponential sweep method for fast measurement of head-related transfer functions," *Journal of the Audio Engineering Society*, vol. 55, no. 7/8, pp. 623–637, 2007.

[10] V. Popa, B. Moroșanu, and C. Negrescu, "Head related transfer function measurement in reverberant rooms," in *Advanced Topics in Optoelectronics, Microelectronics, and Nanotechnologies XI*, vol. 12493, p. 124932I, SPIE, 2023.