AN ALLPASS HEAR-THROUGH HEADSET

Jussi Rämö and Vesa Välimäki

Aalto University School of Electrical Engineering
Dept. Signal Processing and Acoustics, Espoo, Finland

ABSTRACT

In order to create a natural hear-through experience when wearing the headset, the acoustic attenuation of the headset itself must be cancelled. This is obtained by processing the ambient sound signals captured by external microphones. The sound perceived by the headset user will then be a mixture of the ambient sound that leaks through the headset and the processed ambient sound that is reproduced with the headset. We propose a new equalization method for designing such a hear-through system based on an allpass design principle. The proposed method takes the frequency-dependent isolation transfer function of the headset as the input and completes it with an engineered transfer function so that the outcome will be an allpass transfer function with a flat magnitude response.

***Index Terms*—**Acoustic signal processing, audio systems, digital filters, FIR filters

1. INTRODUCTION

Nowadays people are using headphones mostly while commuting and traveling. A noisy environment, such as a bus or a busy street, requires headphones that can effectively attenuate ambient sounds, which has increased the usage of in-ear headphones and active noise control (ANC) headsets [1–4].

The hear-through function in headsets has become common along with the ANC. This is mainly because the ANC attenuates ambient sounds so much that communication is almost impossible without any hear-through implementation. Furthermore, ANC headsets are both active and they often have microphones installed outside the earpiece, which makes the implementation of hear-through function possible [5].

All types of headphones attenuate ambient sounds unevenly in frequency, i.e., the attenuation is frequency dependent. The attenuation curve is basically a lowpass filter [6, 7], which suppresses low frequencies poorly and high frequencies typically too much. It is, however, possible to flatten the attenuation curve using a hear-through function, in which the hear-through signal is highpass filtered ambient noise. This type of a system can be considered to be a kind of a crossover system [8, 9], where low frequencies are transmitted mechanically through the headphone and high frequencies are transmitted electronically via the hear-through function. However, in a headset the leaked ambient sound reaches the ear faster that the processed sound.

The aim of this article is to utilize an allpass filter design method proposed by Strube [10], which allows one to design an allpass filter with a desired beginning of the impulse response. Ideally, when the combined response of the headphone's isolation and the processed hear-through signal has an allpass transfer function, the system executes a perfect hear-through function. Previously, the hear-through function used in augmented reality audio (ARA) has been implemented with three parametric equalizers, both with analog components [11] and digitally [12].

This article presents two different hear-through system designs implemented using two types of headphones, namely supra-aural (on-ear) and in-ear headphones. The supra-aural headphones provide a typical hear-through function, which passes ambient sounds through with no attenuation, whereas the in-ear headphones provide an attenuated hear-through function, which can be used, e.g., in hearing protectors.

This paper is organized as follows. Section 2 describes the allpass filter design. Section 3 presents the headphone measurements for the two headphones. Section 4 focuses on the allpass hear-through systems and their implementations. Section 5 discusses the results and future real-time implementations, and Section 6 concludes the paper.

2. ALLPASS FILTER DESIGN

The allpass filter design used in this paper is based on a design method proposed by Strube [10]. The idea is to extend a given sequence to be an allpass filter. In our case, the given sequence is the impulse response corresponding to the isolation transfer function of a headset. The method is based on the fact that for $N + 1$ real numbers $g_m = g_0, \ldots, g_N$ there is exactly one stable allpass filter $A(z)$ of order N or less, whose impulse response $h(n)$ fulfills $h(n) = g_m$, when $m = n = 0, \ldots, N$ [10].

The allpass filter $A(z)$ is designed as follows: First the known values of g_m are inserted into a matrix

$$\mathbf{H}_{i,j} = \begin{pmatrix} h_{0,0} & h_{0,1} & \cdots & h_{0,N} \\ h_{1,0} & h_{1,1} & \cdots & h_{1,N} \\ \vdots & \vdots & \ddots & \vdots \\ h_{N,0} & h_{N,1} & \cdots & h_{N,N} \end{pmatrix} \tag{1}$$

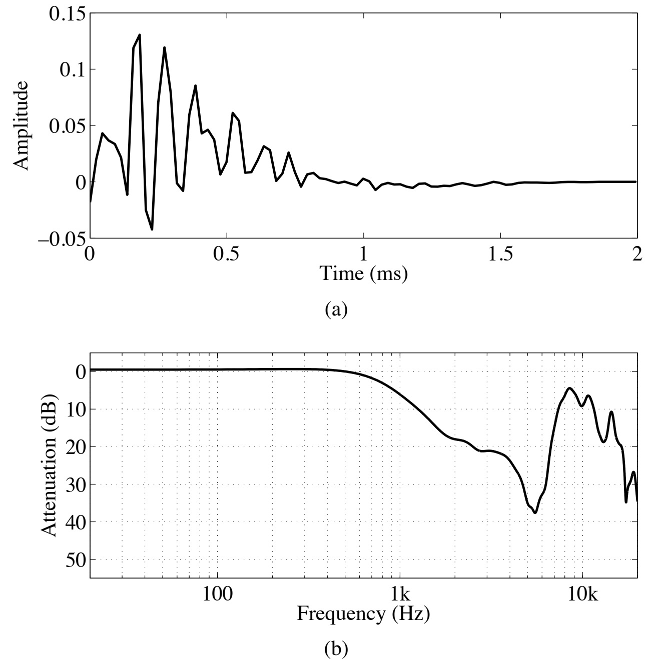

Fig. 1: Measured isolation properties of the supra-aural headphones, where (a) shows the windowed isolation impulse response and (b) shows the magnitude response. The sample rate is 44.1 kHz.

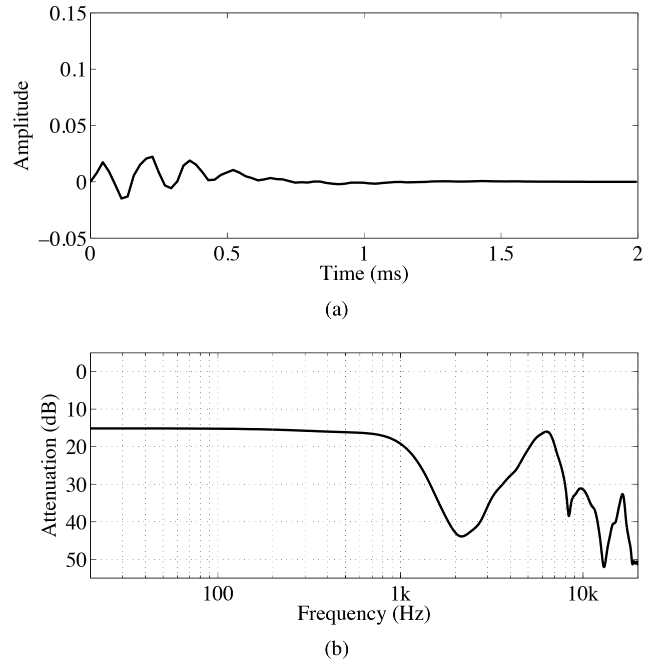

Fig. 2: Measured isolation properties of the in-ear headphones, where (a) shows the windowed isolation impulse response and (b) shows the magnitude response.

according to

$$h_{i,j} = \begin{cases} 0, & \text{when } i+j < N, \\ g_{i+j-N}, & \text{when } i+j \geq N, \end{cases} \tag{2}$$

where $i, j = 0, 1, \ldots, N$. This yields a Hankel matrix

$$\mathbf{H}_{i,j} = \begin{pmatrix} 0 & 0 & \cdots & 0 & g_0 \\ 0 & 0 & \cdots & g_0 & g_1 \\ \vdots & \vdots & \ddots & \vdots & \vdots \\ 0 & g_0 & \cdots & g_{N-2} & g_{N-1} \\ g_0 & g_1 & \cdots & g_{N-1} & g_N \end{pmatrix}, \tag{3}$$

where, by definition, $\mathbf{H}_{i,j} = \mathbf{H}_{i-1,j+1}$. Then, the eigenvalues and the eigenvectors of $\mathbf{H}$ are calculated. Letting μ denote the eigenvalue with the largest absolute value and $\mathbf{a} = (a_0, \ldots, a_N)$ denote the corresponding eigenvector, the transfer function $A(z)$ is composed as

$$\begin{aligned} A(z) &= \mu \sum_{k=0}^{N} a_k z^{-k} \Big/ \sum_{k=0}^{N} a_{n-k} z^{-k} \\ &= \mu \frac{a_0 + a_1 z^{-1} + \cdots + a_{N-1} z^{-(N-1)} + a_N z^{-N}}{a_N + a_{N-1} z^{-1} + \cdots + a_1 z^{-(N-1)} + a_0 z^{-N}}. \end{aligned} \tag{4}$$

This corresponds to an allpass filter whose magnitude response is $|\mu|$ at all frequencies. Note that the value of μ in (4) can also be negative, since the absolute value is only used in the selection of the correct eigenvalue and eigenvector.

3. HEADPHONE MEASUREMENTS

The isolation curves of two different headphones were measured. The headphone types were supra-aural and in-ear headphones. The measurements were conducted in a moderately reverberant listening room with the help of a dummy head. An approximately diffuse sound field was created using four Genelec loudspeakers as well as a subwoofer playing pink noise. The pink noise was recorded using a Brüel&Kjær's head-and-torso simulator (HATS), model 4128C with type 3.3 ear simulator. First the recording was done without the headphones and then again with the headphones. The isolation curve was then obtained as a deconvolution between these two measurements.

Figs. 1 and 2 show the measured isolation properties of the supra-aural and in-ear headphones, respectively. The impulse responses, which have been obtained using the inverse FFT of the isolation frequency responses, shown in Figs. 1a and 2a are windowed using the right half of a Hanning window. The length of both windowed impulse responses is 89 samples. This corresponds to approximately 2 ms, when the sample rate is 44.1 kHz.

Figs. 1b and 2b show the isolation curves of the headphones under evaluation. As can be seen in Fig. 1b, the supra-aural headphones do not attenuate sounds below 500 Hz

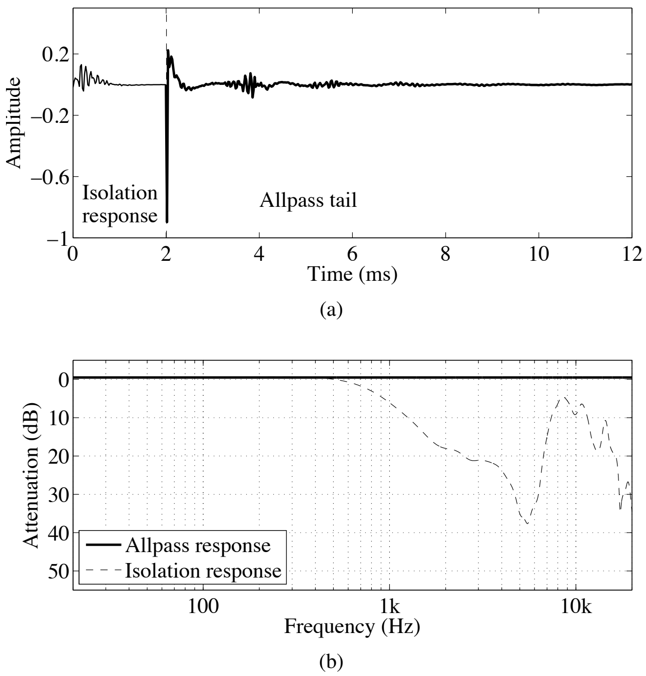

Fig. 3: Allpass equalization of the supra-aural headset, where (a) shows the impulse response consisting of the isolation response of the headphones and the engineered allpass tail and (b) shows the magnitude response of the whole allpass system (solid line) as well as the isolation curve of the headphones (dashed line).

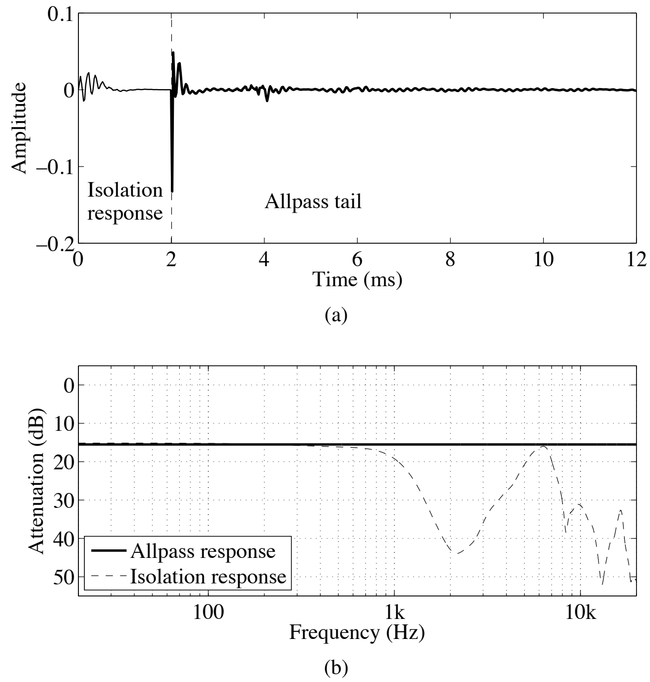

Fig. 4: Allpass equalization of the in-ear headset, where (a) shows the allpass impulse response and (b) shows the isolation curve of the headphones (dashed line) and the magnitude response of the whole allpass system (solid line).

at all, whereas the in-ear headphones in Fig. 2b are shown to attenuate all frequencies more than 15 dB. It should be noted that the use of the type 3.3 ear simulator with HATS often provides a tighter fit for in-ear headphones than that occurring on humans, which can lead to exaggeration in the in-ear headphone isolation measurements at low frequencies (below 1 kHz) [13].

4. ALLPASS HEAR-THROUGH SYSTEMS

The main difficulty in digital hear-through systems is the comb-filtering effect caused by the delay that the processed hear-through signal undergoes [12]. Comb-filtering occurs when a signal is summed to a delayed version of itself. In digital hear-through systems, the sound that leaks through and around the headset can be considered to be the direct sound, whereas the sound that goes through the digital processing can be considered to be the delayed signal. Thus, when these two sounds are summed at the ear drum of the user, it often results in comb filtering. The delay of a commercial DSP evaluation board used in [12] was around 1 ms, without any equalization.

A typical everyday example of the comb-filtering effect is when you listen to a loudspeaker and hear the direct and reflected sound (e.g., from a wall or floor). However, the physics behind the comb-filtering effect in hear-through systems is slightly different from the loudspeaker example. In the loudspeaker case, when the level of the sound reproduced by the loudspeaker is increased, both the direct and delayed sounds are increased in magnitude, whereas in the hear-through case, the direct sound remains constant while the delayed sound can be processed separately. Furthermore, the delayed sound is attenuated according to the isolation properties of the headphones [12].

When a sound propagation of a hear-through system is observed at the ear drum location, the leaked sound arrives first and the processed sound arrives after the delay, which is caused by the digital signal processing (DSP). The delay is caused mainly by the analog-to-digital (AD) and digital-to-analog (DA) converters as well as by the equalizing filter. This delay must be taken into account, since it can cause the comb-filtering effect. Moreover, the isolation response cannot be altered with DSP, and the response of the system can be altered only after the delay that we get from the DSP.

The insight about the independent behaviors of the leaked and delayed sounds enables a novel approach to equalizing hear-through systems by designing the whole transfer function of the system to be an allpass transfer function, as illustrated in Figs. 3 and 4. Figs. 3a and 4a show the impulse response of the whole allpass hear-through system for the mea-

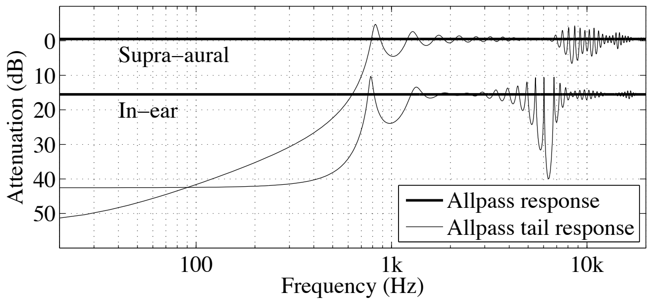

Fig. 5: Magnitude responses of the whole allpass systems (thick lines) and the allpass tails (thin lines).

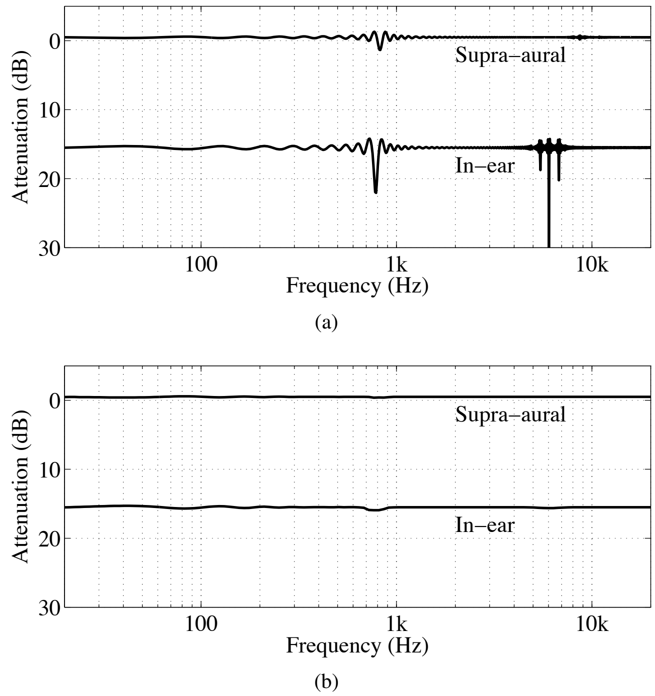

Fig. 6: Truncated allpass FIR filters, where (a) is unsmoothed and (b) is 1/3 octave smoothed magnitude responses of the truncated allpass hear-through systems. The length of the allpass tail is 441 samples.

sured supra-aural and in-ear headphones, where the first part (up to 2 ms) is the same windowed isolation impulse response shown in Figs. 1a and 2a, respectively. The delay caused by the DSP in these examples is 2 ms. The allpass tail is then designed based on the measured isolation response, which is our given sequence g_m, as described in Section 2.

Figs. 3b and 4b show the magnitude responses of the allpass system (solid line) as well as the same measured isolation responses (dashed lines) as shown in Figs. 1b and 2b. As can be seen in Figs. 3 and 4, the allpass design method effectively flattens the isolation response by appending an allpass tail to the actual isolation impulse response measured from the headphones.

Fig. 5 shows the same allpass magnitude responses as Figs. 3b and 4b (thick lines) as well as the magnitude responses of the allpass tails (thin lines). Interestingly, the largest deviations in the allpass tail responses are at frequencies where the isolation curve is near the allpass response, such as around 1 kHz and below 10 kHz (see Figs. 3b and 4b). In other words, the deviations of the magnitude response of the allpass tail are concentrated at those frequencies where the comb-filtering effect would be at its highest. Note that at low frequencies, below about 500 Hz, there is no need for the hear-through signal, and hence the comb-filtering effect does not occur.

4.1. Implementation of the Allpass Tail

The practical implementation of the allpass tail can be dealt with separately. Since the first ‘isolation’ part of the whole allpass response is actually a mechanical lowpass filter caused by the headset, which cannot be changed using DSP, the allpass tail has to be implemented separately. The most straightforward approach is to implement the allpass tail as an FIR filter. However, the constructed FIR filter has to have a reasonable length in practical implementations.

Fig. 6 shows the magnitude responses of the supra-aural and in-ear allpass systems, similarly as shown in Figs. 3b and 4b, but now the allpass tails are truncated to have the length of 441 samples (10 ms). Fig. 6a shows the unsmoothed and 6b shows the 1/3-octave smoothed responses. As can be seen in Fig. 6a, the truncation of the allpass tail introduces ripples to the response, as can be expected. However, Fig. 6b shows the 1/3-octave smoothed response, which is close to the frequency resolution of the human ear at high frequencies [14, 15], and as can be seen, the responses are almost completely flat when the smoothing is used. Furthermore, according to Bücklein [16], single narrow notches, like those in Fig. 6a around 800 Hz, are often inaudible. The chosen length of the allpass-tail FIR filter is a compromise between the sound quality and the computational complexity of the system.

5. DISCUSSION

The proposed algorithm flattens the magnitude response to be at the same level as the maximum of the isolation curve, as can be seen in Figs. 3b and 4b. Thus, when considering the supra-aural headphones, the allpass hear-through equalization could provide a ‘perfect’ hear-through function, e.g., for communication purposes and for ARA [12].

One advantage of the in-ear headphones is their good ambient noise isolation capability. This feature can be utilized in loud environments, such as in live pop/rock concerts [7]. The allpass equalization for in-ear headphones enables a digital implementation of the so called musician’s earplugs, which aim to attenuate all audible frequencies evenly, unlike typical earplugs or hearing protectors. This effect can be seen in

Fig. 4b.

The next step would be to study the requirements of the practical real-time implementation of the allpass equalizer. The practical system is more complicated, since the system includes the frequency responses and delays of the microphone, AD/DA converters, and the headphone driver, which have to be taken into account when designing the allpass tail. The isolation response of the headphone can be longer than the delay caused by the DSP, which means that part of the leaked sound has to be compensated as well. Furthermore, the isolation response can vary when the fitting of the headphones is altered and it is also somewhat dependent on the direction of the ambient sound.

Although there are several aspects to consider when implementing a real-time allpass hear-through system, the authors believe that the system is realizable with the current technology. However, it requires extensive measurements and listening tests to carefully implement the real-time system.

6. CONCLUSIONS

This paper introduced the concept of an allpass hear-through headset. The main issue in hear-through devices is that ambient sounds leak directly through the headset, and thus the leaked sounds reach the ear drum before the digitally processed signal, which undergoes some amount of delay due to the AD/DA converters and processing delay of the DSP. The combination of the direct and delayed signal can cause a comb-filtering effect, which often deteriorates the perceived hear-through experience.

Due to the delay introduced in the DSP, it is not possible to alter the early part of the isolation response. Thus, the idea in the proposed allpass hear-through headset is to design a filter, which leaves the early part of the isolation impulse response untouched and completes the response to be an allpass filter. The paper provided two examples with two different types of headphones. The supra-aural example introduced a flat hear-through response, while the in-ear example showed a flat but attenuated response.

Future work includes real-time implementation of the allpass hear-through headset, as well as the evaluation of the artifact audibility with practical-length allpass-tail FIR filters.

REFERENCES

[1] P. A. Nelson and S. J. Elliott, *Active Control of Sound*, Academic Press Limited, London, UK, 1995.

[2] S. M. Kuo and D. R. Morgan, "Active noise control: A tutorial review," *Proceedings of the IEEE*, vol. 87, no. 6, pp. 943–973, Jun. 1999.

[3] B. Rafaely, "Active noise reducing headset–an overview," in *Proc. International Congress and Exhibition on Noise Control Engineering*, The Hague, The Netherlands, Aug. 2001.

[4] T. Schumacher, H. Krüger, M. Jeub, P. Vary, and C. Beaugeant, "Active noise control in headsets: A new approach for broadband feedback ANC," in *Proc. International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, Prague, Czech Republic, May 2011, pp. 417–420.

[5] S. Priese, C. Bruhnken, D. Voss, J. Peissig, and E. Reithmeier, "Adaptive feedforward control for active noise cancellation in-ear headphones," in *Proc. Meeting on Acoustics, Acoustical Society of America*, Jan. 2013, vol. 18.

[6] J. Rämö, V. Välimäki, M. Alanko, and M. Tikander, "Perceptual frequency response simulator for music in noisy environments," in *Proc. AES 45th International Conference*, Helsinki, Finland, Mar. 2012.

[7] J. Rämö, V. Välimäki, and M. Tikander, "Live sound equalization and attenuation with a headset," in *Proc. AES 51st International Conference*, Helsinki, Finland, Aug. 2013.

[8] S. H. Linkwitz, "Active crossover networks for noncoincident drivers," *J. Audio Eng. Soc.*, vol. 24, no. 1, pp. 2–8, Feb. 1976.

[9] P. Garde, "All-pass crossover systems," *J. Audio Eng. Soc.*, vol. 28, no. 9, pp. 575–584, Sept. 1980.

[10] H. W. Strube, "How to make an all-pass filter with desired impulse response," *IEEE Trans. Acoust., Speech, Signal Processing*, vol. 30, no. 2, pp. 336–337, Apr. 1982.

[11] V. Riikonen, M. Tikander, and M. Karjalainen, "An augmented reality audio mixer and equalizer," in *AES 124th Convention*, Amsterdam, The Netherlands, May 2008.

[12] J. Rämö and V. Välimäki, "Digital augmented reality headset," *Journal of Electrical and Computer Engineering*, 2012.

[13] ITU-T, *Recommendation P.380. Electro-acoustic measurements on headsets*, Series P: Telephone Transmission Quality, Telephone Installations, Local Line Networks. International Telecommunication Union, 11/2003.

[14] D. Griesinger, "Measures of spatial impression and reverberance based on the physiology of human hearing," in *Proc. AES 11th International Conference*, Portland, OR, May 1992, pp. 114–145.

[15] J. Huopaniemi, *Virtual Acoustics and 3-D Sound in Multimedia Signal Processing*, Ph.D. thesis, Helsinki University of Technology, Espoo, Finland, Nov. 1999.

[16] R. Bücklein, "The audibility of frequency response irregularities," *J. Audio Eng. Soc.*, vol. 29, no. 3, pp. 126–131, Mar. 1981.