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Live Sound Equalization and Attenuation with a Headset

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ABSTRACT

An augmented-reality audio application called LiveEQ is described. It captures the ambient sound around the listener with miniature microphones, equalizes the audio signal, and plays it to the listener with headphones. The user hears a mix of the processed headphone sound and the original sound leaking through the headset. The overall sound pressure level at the listener's ear can be made smaller than when listening without headphones. The equalization enables modifying the sound spectrum in a desirable way to improve speech intelligibility or music quality in real time.

1. INTRODUCTION

Live pop and rock concerts have often high sound pressure levels, which forces the audience to use hearing protection. However, typical earplugs attenuate the sound levels unevenly, i.e., the attenuation is highly frequency dependent [1]. There are also more advanced passive hearing protectors available that can reduce sound levels evenly [2, 3]. Moreover, electronic earplugs can provide intelligent gradual sound-level reduction [1].

The proposed live equalization system (LiveEQ) is a novel application for the mobile phone and an in-ear headset: real-time equalization of the sound heard through the earphones. This application provides exciting new possibilities for ordinary users: For example, music heard in a concert through a PA system can be equalized to be more pleasant or more understandable, or speech heard in a noisy environment can be amplified while the background noise is simultaneously attenuated. This technology is based on controlled frequency-dependent amplification of the monaural or preferably binaural microphone signal obtained outside the earphones.

The user of this system can enjoy better sound quality than others in the same environment, although everyone attends the same soundscape. This application has the additional advantage that the user is effectively exposed to less noise, as the equalized signal is played through earphones, which have good noise suppression characteristics. Thus, the processed live sound can be both more pleasant for the user and less dangerous for the hearing.

In contrast to hearing aids, whose aim is to emphasize the frequencies that are not heard naturally, the objective of LiveEQ is to attenuate unwanted or excessively loud sounds and emphasize the wanted sounds according to the user's preference.

Additionally, when an in-ear headset with Active Noise Control (ACN) [4, 5, 6, 7] is used, one can achieve even more pronounced attenuation at low frequencies, giving LiveEQ more control at those frequencies.

This paper is organized as follows. Section 2 describes the LiveEQ system, section 3 presents the Matlab simulation of LiveEQ, section 4 focuses on real-time implementation, and section 5 concludes the paper.

2. LIVE EQUALIZATION SYSTEM

In LiveEQ, ambient sound environment can be equalized in real time. The primary goal of the system is to protect the hearing of a user during a loud concert and at the same time enhance the sound quality of the music. Thus, the user may enjoy a live concert with better sound quality for longer periods of time within safe noise exposure levels.

Figure 1 shows the block diagram of the LiveEQ system. Basically, the perceived sound is the combination of the sound leaked through the headset and the sound that

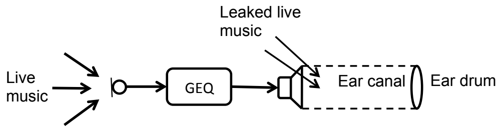


Fig. 1: Block diagram of the LiveEQ, where GEQ depicts a graphic equalizer.

Fig. 2: Headset prototype with microphones attached outside each earpiece.

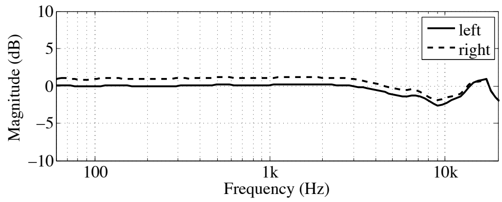


Fig. 3: Frequency response of the microphones used in LiveEQ (1/3 octave smoothing).

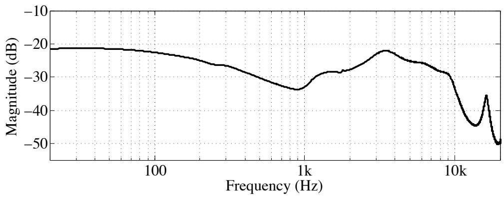


Fig. 4: Frequency response of the WH-701 headphone.

is reproduced with the headset. The system requires a microphone outside the headphones, which captures the sound. The captured signal is processed with a graphic equalizer and played through the earpiece. Noted that a time difference between these two signals can result in a comb filtering effect [8].

2.1. Prototype Hardware

The headset used with the LiveEQ is a prototype built for this purpose. It consists of Nokia WH-701 in-ear headphones and two miniature microphones (Star Micronics MAA-03A-L60), which are attached to both ear pieces, as shown in Figure 2.

The frequency responses of the microphones are shown in Figure 3. As can be seen, the two microphones have quite a flat response, although the levels differ by about 1 dB. Figure 4 shows the frequency response of the Nokia WH-701 measured at the drum reference point (DRP) of Brüer&Kjær's head and torso simulator (HATS) model 4128C with type 3.3 ear simulator. The WH-701 has a perceptually good frequency response, thus providing a good starting point for the LiveEQ processing.

The above mentioned characteristics are important, since they ultimately affect the perceived LiveEQ sound. The sound that finally reaches the eardrum is first recorded with a microphone (with a transfer function H_{mic}). It is then sampled via an analog-to-digital (AD) conversion, equalized with a digital graphic equalizer (having the transfer function H_{geq}), converted back to the continuous domain and reproduced with the headphone (whose transfer function is H_{hp}), and finally transferred to the eardrum through the occluded ear canal (having the transfer function H_{ec}). In addition, the ambient sound also leaks through and around the headset into the ear canal (through the transfer function H_{leak}). Ideally, the sound that reaches the ear drum is

$$Y = (H_{\mathrm{mic}} H_{\mathrm{geq}} H_{\mathrm{hp}} H_{\mathrm{ec}} + H_{\mathrm{leak}} H_{\mathrm{ec}}) X, \qquad (1)$$

where X is the live sound (input) and Y is the LiveEQ sound (output). H_{mic} is depicted in Figure 3, H_{geq} can be designed separately, $H_{\mathrm{hp}} H_{\mathrm{ec}}$ is shown in Figure 4, and H_{leak} H_{ec} is illustrated in Figure 5.

3. MATLAB SIMULATION

A LiveEQ simulator was implemented with Matlab [9] and Playrec [10] based on the headset measurements.

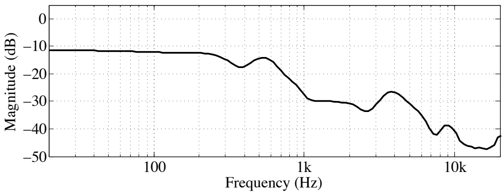


Fig. 5: Isolation curve of the WH-701 headphone (1/3 octave smoothing).

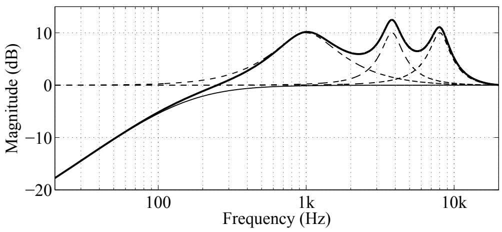


Fig. 6: Magnitude response of the filters, when the gain of the peak filters is set to 10 dB (dashed lines) and the gain of the shelving filter is -30 dB (thin solid line). The thick solid line shows the magnitude response of the whole equalizer.

3.1. Mixer/Equalizer

LiveEQ is equipped with a first-order low shelving filter and with a graphic equalizer [11]. The purpose of the shelving filter is to limit the lowest frequencies, which are usually played back extremely loudly in concerts, since the A-weighting measurement technique allows this. Furthermore, the headset isolates low frequencies the least (only about 10 dB as seen in Figure 5). Thus, in many occasions, the headset is used as a passive ear plug at low frequencies. On the other hand, the attenuation of the headset is usually too strong at high frequencies. A good starting point with the LiveEQ is to design the filters so that the system gives a flat attenuation at all frequencies. This is, however, limited to the lowest attenuation provided by the headset, where, in this case, the flat attenuation range is 0–10 dB.

Due to the good frequency response of the headset, the low shelf filtering gives already quite adequate results. In addition, a three-band graphic equalizer was implemented in order to enable the equalization of live music. The number of the equalizing bands can be easily increased or decreased, and the existing bands can be adjusted in any way. Preset frequency ranges for the three-band equalizer were chosen as follows (see Figure 6):

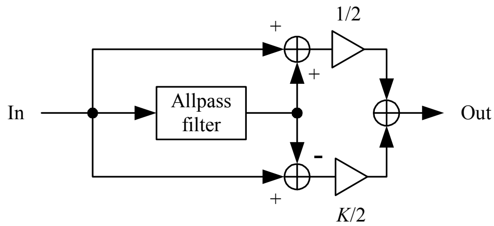


Fig. 7: Allpass equalizer filter structure (adopted from [11]).

1. Midrange – Vocals/Lead,
   approximately 400–2600 Hz
   - This range has the clearest effect on the overall sound.
2. Upper midrange – Presence,
   approximately 2.6–5.2 kHz
   - Too much boost in this range results in harsh and strident sound.
   - Most pleasing sound is achieved by a good balance between this range and the midrange.
3. High end – Brilliance/Cymbals,
   approximately 5.2–10 kHz
   - This range contains mainly harmonics.
   - Too much boost in this range results in harsh and piercing sound.

Both the shelving filter and the peak filters are implemented using the tunable allpass filter structure presented by Regalia and Mitra [11], shown in Figure 7. As can be seen, the transfer function of the filter is

$$H(z) = \frac{1}{2}[1 + A(z)] + \frac{K}{2}[1 - A(z)], \quad (2)$$

where K specifies the gain of the filter and $A(z)$ is the transfer function of the allpass filter.

3.1.1. Shelving Filter

The structure becomes a low shelving filter when $A(z)$ is a first-order allpass filter with the transfer function [11]

$$A(z) = \frac{z^{-1} + a_1}{1 + a_1 z^{-1}}, \tag{3}$$

where a_1 is a frequency parameter [11, 12]:

$$a_1 = \begin{cases} \dfrac{\tan(\Omega/2) - 1}{\tan(\Omega/2) + 1}, & \text{when } K \geq 1, \\[2ex] \dfrac{\tan(\Omega/2) - K}{\tan(\Omega/2) + K}, & \text{when } K < 1. \end{cases} \tag{4}$$

Furthermore, Ω defines the normalized cutoff frequency so that $\Omega = \pi$ corresponds to the Nyquist limit. The cutoff frequency of the shelving filter used in this paper is 5 Hz ($\Omega = 0.000713$).

3.1.2. Peak/Notch Filter

When $A(z)$ is a second-order allpass filter having the transfer function

$$A(z) = \frac{z^{-2} + b(1+a)z^{-1} + a}{1 + b(1+a)z^{-1} + az^{-2}}, \tag{5}$$

the structure becomes a peak/notch filter [11]. The frequency parameter a for a peak ($K \geq 1$) and notch ($K < 1$) filter, respectively, is defined as [11, 12]

$$a = \begin{cases} \dfrac{1 - \tan(\Omega/2)}{1 + \tan(\Omega/2)}, & \text{when } K \geq 1 \\[2ex] \dfrac{K - \tan(\Omega/2)}{K + \tan(\Omega/2)}, & \text{when } K < 1 \end{cases} \tag{6}$$

where Ω denotes the normalized filter bandwidth. The parameter b is

$$b = -\cos(\omega_0), \tag{7}$$

where ω_0 is the normalized center frequency [11].

The center frequencies and bandwidths of the preset filters are given in Table 1, where

$$f_c = 2\pi\omega_0/f_s \tag{8}$$

is the center frequency in hertz,

$$BW = 2\pi\Omega/f_s, \tag{9}$$

	f_c (Hz)	ω_0	BW (Hz)	Ω
Peak 1	1000	0.1425	800	0.1140
Peak 2	3800	0.5414	1000	0.1425
Peak 3	8000	1.1398	2000	0.2850

Table 1: Filter parameters, $f_s = 44.1$ kHz.

is the bandwidth of the filter in Hertz, and f_s denotes the sampling rate, which is 44.1 kHz throughout this paper.

3.2. Matlab Graphical User Interface

Figure 8 shows a screen shot of the graphical user interface of the LiveEQ simulator implemented with Matlab. The gain of the three equalizer filters is adjusted using sliders labelled ‘Vocals/Lead’, ‘Presence’, and ‘Brilliance/Cymbals’. The thick black curve illustrates the passive isolation of the WH-701 headset (i.e., the leaked sound), the thin black curve illustrates the frequency response of the equalizer controlled with the sliders (i.e., the sound reproduced with the headphone), and the red curve is the spectrum of the actual perceived LiveEQ sound, i.e., the time-domain sum of the leaked sound and the sound reproduced with the headset.

There are three listening options in the simulator:

- Open ears – Unprocessed signal.
- Ear plugs – Simulates the situation where the headset is used as ear plugs.
- LiveEQ – Simulates the LiveEQ, i.e., the sum of the leaked and the processed sound.

Furthermore, a ‘bass boost’ function is implemented to simulate the excessive amount of low frequencies in the music, which is often the case in pop or rock concerts, especially indoors. The bass boost is implemented using the Regalia-Mitra shelving filter, described in Section 3.1.1.

4. REAL-TIME IMPLEMENTATION

A real-time LiveEQ was implemented using an Analog Devices DSP evaluation board (EVAL-ADAU1761Z). Implement a real-time demonstrator using a regular sound card and Matlab is not possible, since the sound card is primarily designed to playback and record signals, where the delay is not critical. For example, a

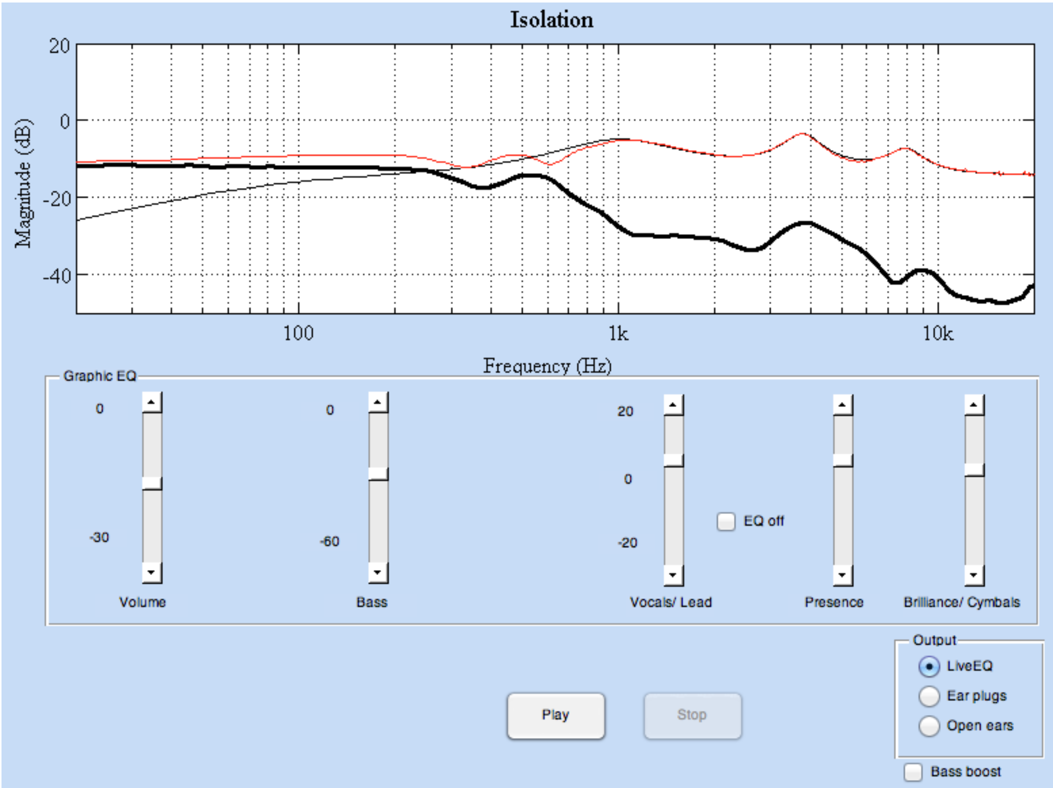


Fig. 8: LiveEQ Matlab simulator. The thick black curve illustrates the passive isolation of the WH-701 headset, the thin black curve illustrates the frequency response of the equalizer, and the red curve is the actual perceived LiveEQ.

MOTU UltraLite mk3 audio interface [13] has a constant delay of approximately 20 ms, which is unacceptable in the LiveEQ application. The real-time implementation requires external sound sources and the actual built headset prototype.

The DSP board was used due to its low-latency performance. The measured throughput delay time of the DSP board is between 0.95–1 ms, which is mainly caused by the AD and digital-to-analog (DA) converters [8]. A 1-ms delay is acceptable and allows the usage of the DSP board as a real-time LiveEQ processor.

As can be expected, the implementation of the equalizer filters slightly increases the group delay of the system. The group delay increases at the stop band of a shelving filter as well as around the peak of a biquad filter. A notch filter typically decreases the group delay, because the phase slope is positive around the center frequency of the notch.

The preset frequencies for the graphic equalizer are the same as in the Matlab simulation. Figure 9 shows the measured frequency response of the equalizer implemented with the DSP board when all three filters are adjusted to have a gain of 10 dB.

Figure 10 shows the measured group delay of the DSP board when all three filters are set to 10 dB. The group delay was measured by playing a sine sweep through the

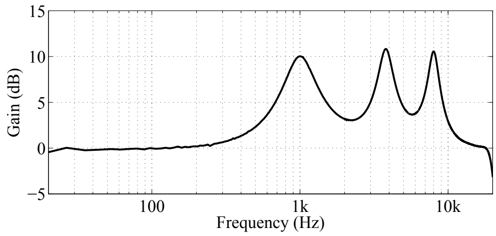


Fig. 9: Frequency response of the graphic equalizer measured from the DSP board when all three filters are set to 10 dB.

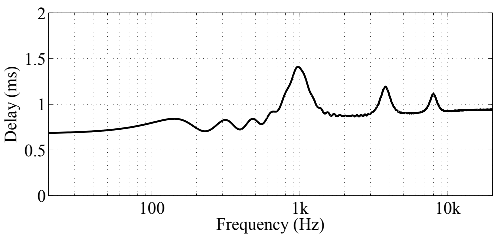


Fig. 10: Group delay of the DSP board when all three filters are set to 10 dB.

DSP board. A MOTU UltraLite mk3 audio interface was used, since it has a constant delay which could be measured and removed from the results.

4.1. Comb Filtering Effect

Adding a delayed copy of a signal to the signal itself results in a comb filtering effect [14]. In the case of LiveEQ, the latency of the processed sound should be small, because it is added to the leaked sound, which has only gone through the headset to the ear drum.

The measurement in Figure 10 suggests that the delay of the real-time LiveEQ processing is below 1.5 ms. Furthermore, the idea in LiveEQ is not to reproduce low frequencies with the headset, since the low frequencies are usually played very loud in live concerts. Thus, the comb filtering effect does not occur at low frequencies. Moreover, the attenuation of the headset increases toward high frequencies, which effectively decreases the comb filtering effect (see Figures 5 and 12(b)).

The comb filtering effect can be illustrated by using an FIR comb filter [14], which in this case has the transfer

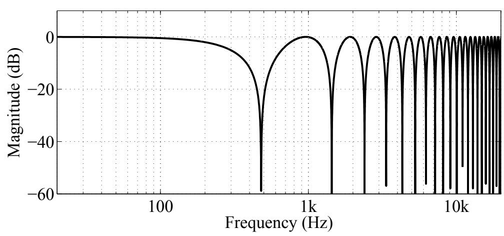


Fig. 11: Comb filtering effect implemented with FIR comb filter, when the signals have the same energy and the delay is 1 ms.

function

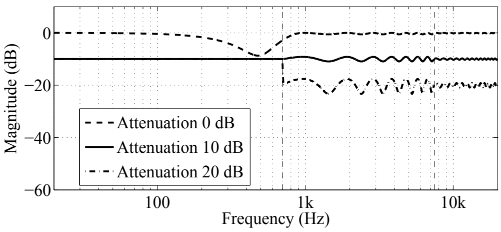
$$H(z) = g_{\mathrm{d}} + g_{\mathrm{eq}} z^{-L}, \tag{10}$$

where g_{d} is the gain for direct sound (i.e., the leaked sound), g_{eq} is the gain for the sound reproduced with the LiveEQ headset, and L is the delay of the reproduced sound in samples.

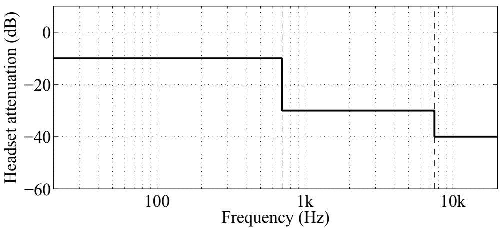
Figure 11 shows the worst-case scenario where the direct and delayed sound has the same amount of energy ($g_{\mathrm{d}} = g_{\mathrm{eq}} = 0.5$) and $L = 45$ samples, which equals to a 1-ms delay.

Figure 12(a) shows the amount of the comb filtering effect for different constant attenuations of LiveEQ with a delay of 1 ms. The desired response in each case is a flat magnitude response. The headphone attenuation is simplified to that in Figure 12(b) so that the attenuation is 10 dB below 700 Hz, 30 dB between 700 and 7500 Hz, and 40 dB above 7500 Hz.

As can be seen in Figure 12(a), when the target attenuation is 0 dB, the main comb dip is at around 500 Hz, where the attenuation of the headset is at its weakest. At frequencies above that, the sound originates mainly from the headphone due to the good attenuation of the headset, and thus the comb filtering effect is almost nonexistent. On the other hand, when the desired attenuation is 10 dB, the comb filtering effect does not occur at low frequencies at all, since the headphone does not reproduce any sound in this range. Furthermore, for a desired attenuation of 20 dB, the headset cannot attenuate low-frequency sounds as much, and the attenuation is still 10 dB. However, at frequencies above 700 Hz, a 20-dB attenuation is achieved. As can be seen, the comb filtering effect increases at high frequencies compared to the 10



(a)



(b)

Fig. 12: Theoretical comb filtering effect in LiveEQ with a delay of 1 ms, where (a) illustrates the amount of comb filtering when the liveEQ is adjusted to have different constant attenuations and (b) presents the simplified attenuation of the headset.

and 0 dB attenuations, since the energy of the sound originating in the headphone approaches that of the leaked sound.

4.2. Discussion

The real-time implementation of LiveEQ may require an external headphone amplifier, since the digital amplifiers on the DSP board induce whistling in the signal, implying that the amplification capability of this particular DSP board is weak. Any external amplifier used should have as a low delay as possible; an analog device is preferred.

Furthermore, the passive attenuation of a headphone behaves as a lowpass filter. Thus, the leaked sound has to propagate through the passive mechanism (i.e., a mechanical lowpass filter having the response shown in Figure 5), and while doing that it is subjected to an additional group delay [15]. According to Rafaely [15], the group delay of the mechanical lowpass filter is approximately 0.7–1.7 ms at low frequencies. Thus, the mechanical group delay allows additional electronic delay in the system, caused by the digital filters and AD/DA converters, without reduction in the performance of LiveEQ.

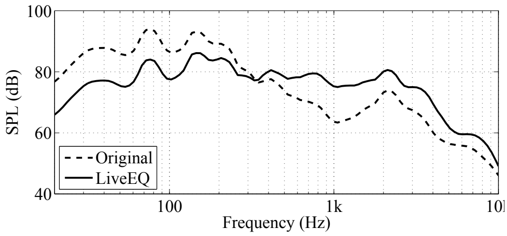


Fig. 13: Frequency response of a 7-second music clip (1/3 octave smoothing), where the dashed line is the original spectrum and the solid line is the spectrum after the LiveEQ processing. The gains for the equalizer were -30 dB for the shelving filter, and 18 dB, 10 dB, and 10 dB for the midrange, upper midrange, and high end peak filters, respectively.

Figure 13 shows an example case where a 7-second bass-heavy music sample (dashed line) is equalized to have a flatter frequency response with the LiveEQ system (solid line).

5. CONCLUSIONS

This paper described a novel augmented-reality application called LiveEQ, which equalizes the ambient sound around the listener in a desirable way in real time. The main purpose of use is in loud concerts, where hearing protectors are required. Normally when one uses ear plugs, the sound is attenuated too much at mid and high frequencies. With LiveEQ, mid and high frequencies are allowed to pass through the headset in order to enhance the music quality while still using hearing protection. This way users can reduce their noise exposure and still enjoy the concert with pleasant sound quality.

LiveEQ was implemented using Matlab and Playrec together as well as with a DSP evaluation board. The Matlab/Playrec implementation serves as an easy-to-setup demonstrator tool, where the live concert environment is simulated, whereas the DSP board implementation serves as an actual real-time prototype of LiveEQ. The real-time implementation was realized with the DSP board, since it has sufficiently small delay. The simulations and the real-time prototype indicate that LiveEQ is realizable and provides good sound quality in concerts while still protecting the hearing of the user.

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