DIGITAL FILTRATION OF AN AUDIO SIGNAL TO MINIMIZE THE RESONANCES IN THE ROOM ACOUSTIC RESPONSE

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Abstract:

The paper presents an alternative method of suppressing audio resonance in the room acoustics, which has a major influence on the quality of audio reproduction. The method uses feed-forward technique filter to suppress the excitation of resonance. The basic algorithm was originally developed for robotics and positioning system applications and is adapted for the acoustic area.

Introduction

The quality of an audio signal reproduction in the room is depending on many factors. Since the CD audio has become standard, the audio devices have to provide high quality reproduction. In comparison with video playback, where the lower resolution and frame rate is often satisfactory for the user, very high standards are expected in audio products area. High performance of the modern consumer electronics assures the sufficient quality of the digital-to-analogue conversion even in the lower-price segment of the players. The differences between various products are brought on especially by terminal amplifiers. The very significant influence on the audio reproduction have loudspeakers and acoustical qualities of the audition room of which the reflections and resonance are most problematic. Overall audio device quality is the matter of selection of the appropriate product, the improvement of a room response is then the ultimate task. The problems with resonance appear also in many other disciplines, we will investigate feed-forward techniques for their suppression which are applied in positioning and robotics.

Room acoustic

Room acoustic is influenced by a large amount of factors: the surface and geometry of interior and objects in it, position and orientation of the listener, position and orientation of loudspeakers, presence and position of persons, air temperature and humidity. The main problem in the room environment is the interior variability. It is particulary important in small audition rooms (especially in cars - even small changes, such as the presence / absence of people, can cause a major change in the room transfer function).

The methods for improvement of acoustic and aural quality of rooms have waste possible application areas. Simpler applications provide suppression of selected unsuitable acoustic effects, such as reverbration. This is usual especially in large auditoriums, stadiums and churches. Many auditoriums require the precise reproduction and therefore it is needed to compensate the acoustic transfer function of the room.

On the other side, different models are developed for design and testing of rooms for simulation of different environments. Here the reflection and resonance of for target space...
(projected buildings or cars) are simulated using artificial reverbrators. The modeling and simulation of room acoustic helps to optimize the topology of room and loudspeakers as well as to adjust the parameters of the reproduction chain.

The research is made also in the direction of human perception and analyse on listener requirements. People usually prefer to listen the audio signal with reflected side channels, this is employed in architecture for the design of auditorium halls, on the other side, too many reflections and the prolongation of echo is distorting the quality of audition\textsuperscript{1}.

**Room reverberation and resonance**

The room response can be divided into the direct sound (which has the major effect on the perceived direction), early reflections (which have an effect on detailed spatial recognition), late reflection clusters and late reverbation (sound envelopment). Especially the late reverbation and resonance can have negative effect on the quality of the audio reproduction. Therefore the resonance modes of the room are the subject of our study.

\[
F_{\text{ref}} = \sum b_i z^{-i}
\]

\[
A_{\text{res}} = K |A(j \omega_{\text{room}})|
\]

Fig. 1. Sketch of the sound source and the listener in the room

Fig. 1 shows the propagation of sound signals in the room with the listener. Statistically, for a regular room, the density of mode frequencies (modes/Hz) at mid to high frequencies is\textsuperscript{[2]}:

\[
N_{\text{mode}} (f) = \frac{4\pi f^2}{c^3}
\]

and the reflection density (reflections/s) for late reverbation is\textsuperscript{[2]}:

\[
N_{\text{reflections}} (t) = \frac{4\pi c^3 t^2}{V}
\]

Overall room response can be modeled by the linear-system transfer function (sometimes referred as a IIR model – Infinite Impulse Response):

\[
G(s) = \frac{b_M s^M + \cdots + b_1 s + b_0}{a_N s^N + \cdots + a_1 s + a_0}
\]

The discrete form:
The simpler FIR model (Finite Impulse Response) uses only the numerator of (4):
\[ G(z) = d_0 + d_1 z^{-1} + \ldots + d_M z^{-M} \]  \hspace{1cm} (5)

The IIR model is complex, however the numerical identification with ARX, ARMA, OJ, BJ.. models can be applied here. The FIR model is simple, but too long function is necessary for the real system modeling. More complex model is a structurally organized set of fixed-pole IIR filters - Kautz Filters²[2][3].

The research is oriented on the suppression of the resonance in the high reverberant room. Such room has a transfer function, which characteristic polynomial contains a complex pair of poles for each resonance mode. The transfer function with one mode of vibration (natural frequency \( \omega_0 \) and damping \( b \)) has the form⁴:
\[ G(s) = \frac{K\omega_0}{s^2 + 2b\omega_0 s + \omega_0^2} \]  \hspace{1cm} (6)

Since the acoustic features of the room depend on its construction and cannot be changed, the way to eliminate the resonance is to prevent its excitation. Since the system (room) acts as an amplifier of harmonic signals with frequency near to its vibration modes, the solution is to remove these from the reproduced signal. Considering the signal should not be changed in the great manner and be as similar to the original one as possible, it is necessary to remove very thin parts of the signals spectrum components.

**Standard filtration of the signal**

Recent processors used for signal processing have usually enough computational power to perform quite simple tasks for filtration. Possibility of an online identification and an adaptation on the other side would require quite complex computations and is hardly realizable. Important criteria is the overall phase change, it should remain small.

For the audio signal filtration are usually used notch filters. The identification of resonance frequencies is realized offline and the result system is justified for the particular room. The notch filter transfer function (with frequency \( \omega_0 \) and quality factor \( Q \)) is:
\[ G(s) = \frac{s^2 + \omega_0^2}{s^2 + s\omega_0 / Q + \omega_0^2} \]  \hspace{1cm} (7)

The room online identification using cross correlation / Hadamard Transformation is very computationally demanding, therefore is not used in commercial systems.

**Digital filtration of the signal by an optimized filter**

An alternative to the serial set of band-pass or notch filters is an optimized digital filter. In comparison to the band-pass filter, the proposed digital convolution filter has an exact frequency of effect (like the notch filter). The disadvantage of this quality is the increased
sensitivity on the parameters, especially the frequency of resonance. It shows high efficiency and simple design. The main idea behind this filter is the convolution of the signal with the specially designed sequence of Dirac impulses. Set of Dirac impulses with amplitudes $A_i$ and time delays $\tau_i$ can be expressed as follows (the later is in transformed form):

$$D(t) = \sum_i A_i \delta(t - \tau_i)$$

$$F_D(s) = \sum_i A_i e^{-s\tau_i}$$  \hspace{1cm} (8)

We can represent the Dirac impulses as vectors in the phase plane ($2\pi$ rad corresponds the period of natural oscillations). The scalar sum condition and the vector sum condition:

$$\sum_i A_i = 1, \quad \sum_i A_i e^{-j\omega\tau_i} = 0$$  \hspace{1cm} (9)

ensure the filtration of the $\omega_0$ frequency. Fig. 2 shows the application of the ZVD filter\cite{4}, the filter affects only the narrow area around the filtrated frequency $f_0$.

![Fig. 2: The sin() audio signal generated f=1000Hz, filtrated with filter $f_0=800$Hz (left), the spectrum of signal (f=1000Hz, $f_0=800$Hz, middle) and the spectrum of the filtrated signal with frequency equal to filtration frequency (f=800Hz, $f_0=800$Hz, right).](image)

The numerically optimized set of impulses can ensure the filtration of the signal on several frequencies, very important advantage is the smaller time delay of the filter in comparison to the set of notch filters and therefore smaller degradation of the original audio signal.

**Conclusion**

High demands on quality of audio reproduction is increasing requirements on compensation of effects of the audition room, especially problematic is the resonance. Classical approach to suppress the excitation of resonance includes the application of notch filters. We are proposing the application of advanced methods of control of mechatronical systems with numerically optimized digital filters designed for resonance suppression.

**References**