Paper ID: E&TC17

REVIEW OF LOUDSPEAKER EQUALIZATION

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Abstract— Different methods to design digital filters aimed for equalization of loudspeaker/room responses are considered. Design of inverse filters is based on measured loudspeaker/room impulse responses combined with room- and psychoacoustic knowledge. Frequency dependent smoothing and nonlinear equalization effort is applied, and a new iterative method has been proposed.

Index Terms— Deconvolution, MIMO, FIR, IIR, Kautz and Wrapped filter

I. INTRODUCTION

Loudspeaker equalization is an essential technique in audio system design. As loudspeaker arrangement driven through a compensating network. The compensating network has an amplitude-frequency response characteristic which is substantially flat over the range of frequencies for which the loudspeaker characteristic is flat, and which rises at higher and lower frequencies to compensate for the decline in the loudspeaker characteristic at those frequencies. The compensating network described however, does not compensate for the periodic variations in response exhibited by many loudspeakers at the higher frequencies.

It is an object of the present invention to extend the frequency range over which a loudspeaker can be used to provide high fidelity reproduction. It is a more specific object of the present invention to provide a loudspeaker arrangement in which variations in the amplitude-frequency response characteristic resulting from reflections and/or mechanical resonances (i.e. regardless of origin) are reduced or substantially eliminated. To these ends, the present invention provides speaker equalization (i.e. correction of speaker response to obtain a substantially flat frequency-amplitude response characteristic) through incorporation of discrete time filter means in the input to the speaker.

II NEED OF EQUALIZATION

A. Low frequency room modes

At low frequencies it is recommended to have a complete correction of the room modes (caused by standing waves) including high-Q room resonances; but this will not work at higher frequencies over a reasonable listening area. Several musical sounds contain only few spectral components(with substantial separation), and we have only a low density of room modes below the Schroeder frequency (typical around 100 - 200 Hz), therefore the need for equalization in this low frequency region is very pronounced.

B. Room- and psychoacoustic criteria

As mentioned, it is not possible to make a complete deconvolution across the listening area. Fortunately, the target for the equalization task is not an anechoic chamber. Perception of distance is poor in anechoic surroundings, the perceived distance will depend directly on the play-back level. Further, the spatial impression is poor in a standard two channel stereo configuration, without side wall reflections. In fact, it has been proposed to improve the listening conditions in a typical living room by adding more lateral energy, see Griesinger [1]. Therefore we have to decide to what degree the reverberant sound field shall be reduced and how we shall equalize the direct sound and the early reflections. Shall we define different target rooms (application specific) described by the common room acoustic parameters like reverberation time, clarity etc? In this case the optimal target room will depend on the type of music. A reverberation time around 0.4 sec. seems to be an appropriate target since much CD material is mixed to sound good in such surroundings. What levels of early reflections are acceptable? If these levels cannot be achieved, can we then improve the sound quality by adding some early reflections combined with a suppression of the most dominant early reflections? Other fundamental issues, related to psychoacoustic criteria are, how flat a frequency response do we need, and what is the optimal equalization of peaks/dips.

C. Phase equalization

A fundamental issue is: Can we ignore equalization of the excess phase part in loudspeaker/room transfer functions? At the moment there is no clear answer, in an earlier investigation we have shown that the excess phase, under certain circumstances, is audible, see Johansen & Rubak [2]. It is not clear how important it will be to separate the two parts of the compound impulse response for loudspeaker/room. Craven & Gerzon claim that it is important to compensate the phase response of the woofer unit, and propose a linear phase response. A recent review

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concerning equalization of loudspeakers is given by Karjalainen et al. [11]

III. EQUALIZATION OF LOUDSPEAKER AND ROOM

A. Introduction

For small rooms (including normal living rooms) we have two very important issues, how to correct the early reflections (maybe combined with the direct sound) and high Q low-frequency room resonances. The reverberation time is usually in the range 0.4 - 0.8 sec therefore the need for a reduction is often limited. The value (in this range) of the reverberation time is less important than the temporal distribution and the levels of the early reflections.

B. Loudspeaker equalization

It is difficult to separate loudspeaker/room impulse responses. It is possible to derive an impulse response which is close to the effect caused by the loudspeaker alone (position independent part). The simplest way is to use an anechoic measurement e.g. in 30 degree of the loudspeakers impulse-response. The directional properties are of cause not included, but the loudspeakers basic frequency response is accounted for. Craven & Gerzon [3] propose an equalization of the loudspeaker including non-minimum phase correction. In their opinion it is important to achieve a linear phase characteristic for the woofer high pass response. A high quality loudspeaker is mainly a minimum phase system, but the crossover network can include a non-minimum phase part (all pass). The impulse response for that part is short (a few ms), therefore it is possible to correct the all pass part using a reasonable short delay necessary to obtain a causal impulse response. As pointed out above, it seems to be the most appropriate for small rooms to equalize the combined impulse response for loudspeaker/room.

C. Room equalization

Room impulse responses are generally non-minimum phase systems see Neely &Allan [4], Johansen & Rubak [2]. Rooms with very short reverberation times seem to be closer to minimum phase. But what about the very important early reflections, are they minimum phase? Genereux [5] discusses this issue using a simple model. He consider a direct sound combined with one broad-band reflection (frequency independent reflection coefficient). The transfer function is given by:

$$H(z) = 1 + a z^{-m}$$

In this case all m zeros are placed on a circle with radius a 1/m, and therefore we have a minimum phase system for $a < 1$. In other words, if the amplitude of the reflection is less than the direct sound (this condition is fulfilled for ordinary rum) we have a minimum phase system. This is not a general proof of the hypothesis that all early reflections represent a simple minimum phase system, but data presented by Mourjopoulos [6] show that non-minimum phase components in a measured room impulse response are predominantly in the reverberation tail. There is some evidence, according to Craven & Gerzon [3], that the low-frequency part of the room impulse- response also is close to minimum phase. A minimum phase equalizer seems to be appropriate for equalization of both the early part of the room impulse-response and the low-frequency high Q resonances.

D. Frequency resolution

Equalization of low-frequency high Q room resonances requires a very high frequency resolution, about 1-2 Hz. Implementation of this resolution, using an FIR filter requires an unrealistic number of filter coefficients, in the order of 40,000 taps. Craven & Gerzon have solved this problem by using down-sampling. An alternative method is application of “Warped-filters”, Johansen & Rubak [7].

IV. EXISTING WORK

The existing works on loudspeaker equalization can be classified into several categories: Deconvolution method, Use of wrapping filters, MIMO feed forward control. This are listed below.

A. Convolution Method:

Zhang Ping [8] proposed loudspeaker equalization is an essential technique in audio system design. A well-known equalization scheme is based on the deconvolution of the desired equalized response with the measured impulse response of the loudspeaker. In this paper, a post-processing scheme is combined with the deconvolution-based algorithm to provide a better equalization effect. Computer simulation results are given to demonstrate the significant improvement that can be achieved using this method.

B.Wraped filter Design:

Matti Karjalainen [9] proposed a technique based on wraped filters. They allow for the design of equalizers on non uniform frequency resolution that is characteristic to auditory perception, which enables also to use lower filter orders which compensates for inherently more complex structures of wrapped filters.

The proposed wrapped structure requires less precision, avoid excessive emphasis on equalization of high frequency resonances and antiresonaces which easily happens with uniform frequency.

C.MIMO feed forward control:

Adrian Bahne. [10] In this work we presented a method for loudspeaker-room equalization by means of combining a general MIMO equalization method presented by the authors earlier together with a novel pair wise channel similarity criterion. The similarity criterion is motivated by the requirements of multichannel standards like stereophonic or 5.1 surround sound reproductions, where phantom images are created based on amplitude and phase differences between symmetric channels. Correct playback of recordings using these techniques, basically all multichannel recordings, thus requires symmetrical and

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therefore similar RTFs. To assess the proposed method a measure of RTF similarity is required. To this end we introduced the cross-correlation between two channels in narrow frequency bands corresponding to the critical bandwidth of the auditory filter. The proposed method was then investigated by means of measurements of two multichannel audio systems

The problem of loudspeaker response equalization is simpler than the correction of a full acoustic path including room acoustics. Loudspeaker impulse responses are relatively short and the magnitude response is regular in a well designed speaker. EQ filter techniques proposed for the purpose include FIR filters, warped FIR and IIR filters [11], and Kautz filters [12].

While flattening of the magnitude response also in this case is relatively easy to carry out, difficult problems are found particularly in reducing excessive reverberation, reflections from room surfaces, and sharp resonances due to low-frequency room modes. Reduction of the effect of perceived room reverberation, in order to improve clarity, is a very hard task because of the highly complex modal behavior of rooms at mid to high frequencies. By proper shaping of the temporal envelope of the response, for example, by complex smoothing technique in EQ FIR filter design [13, 14], this can be achieved to some degree. This requires necessarily high-order equalization filters. Counteracting room surface reflections is only possible to a specified point in the space, from where the receiver is allowed to move less than a fraction of wavelength of the highest frequency in question. At lowest frequencies, modal equalization [15] has been developed to control the temporal decay characteristics of modal resonances that have too high Q-values.

V OBJECTIVES OF THE STUDY

To control excessively long decays is problematic or difficult with conventional passive means. Also develop the model decay behavior of a loudspeaker-room system. To design, development and analysis FIR filter for equalization and implement and performance evaluation of reconfigurable FIR filter using FPGA.

By fulfilling the above mentioned objectives, develop a model for loudspeaker and room equalization.

VI CONCLUSION

Design criteria for equalization of loudspeaker/room impulse responses are developed. The criteria are based on a systematic analysis of the equalization scenario including loudspeaker, room and listener. The analysis includes room acoustic and psychoacoustic factors, as well as theoretical aspects of time- and frequency domain analysis. The inherent problems considering equalization of non-minimum phase systems are discussed. Because a complete deconvolution is impossible to achieve across a reasonable listening area we discuss different possible targets for the equalization task. Focus is put on the problematic position sensitivity, which is very severe at mid to high frequencies. Averaging across the listening area is one approach, but we have chosen a alternative method based on decreasing frequency resolution at higher frequencies. Different preprocessing techniques are considered. Optimization is based on MATLAB simulations, and evaluation of the corrected impulse responses is based on a new software toolbox. The equalizer is based on measurement in one or 4 listening points of the compound transfer function for loudspeaker/room/listener, and minimum phase EQ design. A new method is under investigation. The distribution of early reflections is modified by adding reflections, to obtain a more random distribution and a better balance in relation to the reverberant part. This procedure is combined with the previously used frequency domain techniques.

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