FIR 필터를 사용한 청취 환경 보정 시스템

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FIR ROOM RESPONSE CORRECTION SYSTEM

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ABSTRACT

Due to advances in electronics very high quality audio reproduction is today possible. But the listening environment causes deviation of the audio system from the expected behavior. Firstly the listening Room significantly changes the audio signal frequencies and their phase. Secondly the position of the user in the room affects the perceived sound. With existing DSP technology it is possible to adequately correct these effects. In our work we developed a room correction system, correcting up to 7.1 channels using dual Motorola 56367 fixed point DSP's, implementing position dependent room effects measurement, real time compensation filter design and equalization filtering procedures.

1. INTRODUCTION

The effects of the room on acoustic waves traveling have been well studied [1]. Acoustic waves travels in the air, being absorbed, traversed and reflected by walls and objects until they reach the listener. Due to this some frequencies get amplified and others get attenuated, along with reverberation effects. These distortions introduced by the room significantly distort the sound at the listener position and over a period of time various methods have been proposed to improve the system performance.

The first attempt at solving the problem involved passive methods such as finding suitable loudspeaker positions in a room to minimize distortions at the listening position [2]. Subsequently, a number of active sound control solutions have been proposed to correct the above mentioned distortion effects [3-4]. The proposed solutions typically introduce an equalization filter in the signal path, which is an inverse of the room response measured at the listening position. The initial step in the solutions proposed is to use a test signal to measure the room characteristics at the listening position, design an equalization filter based on the results of measurement and equalize the audio signal before it is passed through the distortion path. Test signals might also be used to measure inter speaker position related effects.

A major challenge in the active signal processing techniques is the design of the equalization filters. Equalization filter design problems involve the non-invertibility of room response, the response being non minimum phase. Non minimum phase room responses have zeroes outside the unit circle and hence their inverse filters are unstable, there poles being outside the unit circle. Inverse filter design techniques for non minimum phase systems using the regularization principle [5] have been recently introduced to solve the problem.

Challenges faced by the proposed solutions also include the sensitivity of the room response to position. A few multi point room equalization solutions have been proposed which typically involve using an averaged frequency response at multiple points in the room and equalizing the averaged/clustered response [6-7]. The proposed method does improve the distortion at multiple user locations but not be worth the added complexity and increased measurement time.

The type of filter used in the equalization filter is a topic of debate. IIR filters have been used in low complexity systems but they introduce phase errors which are very annoying particularly at low frequency. Using IIR severely limits the low frequency performance of the EQ systems. Typically multi band band IIR filters or parametric EQ systems are used in IIR implementations [8-9].

It is known that differences in amplitude and time between different speakers can introduce spatial cues in the sound [10]. The multiple loudspeaker system hence should be corrected for inter speaker volume gain and time differences because of the user position with respect to different loudspeakers.

The following paper describes the real time room effects measurement and compensation system developed by us. We propose solutions to the above mentioned problems focused towards a simple real time implementation with high quality. The designed system is able to perform measurement and filter design on the DSP hardware itself instead of requirements of a PC [11], and uses dual Motorola 56367 Audio DSPs. Section 2 describes in detail the algorithm used in our implementation. Section 3 describes the DSP processor implementation details followed by results of our work and references in sections 4 and 5.

2. MEASUREMENT ALGORITHM DETAILS

Figure 1 shows the block diagram of the measurement algorithm. The subsequent sections describe the algorithm in detail.

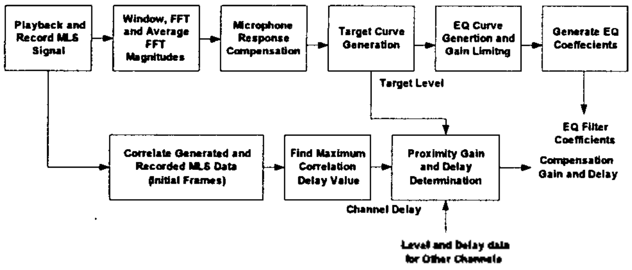

Figure 1. Room Correction System Block Diagram.

2.1. Room Response Calculation

The first step in the room correction process is to characterize the audio signal received at the user position and determine the effects of the room and speaker configuration mathematically by passing a test signal through the loudspeakers using the configuration shown in figure 2.

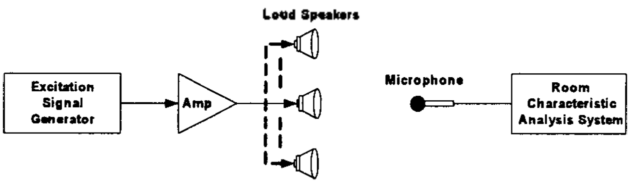

Figure 2. Room characteristics measurement configuration.

We generate a maximum length sequence (MLS) [12] of sufficient periodicity for the room excitation signal. The recorded MLS signal is framed using overlapping frames of a suitable frame length and windowing is done. The data frames are then FFT transformed using a FFT length same as the data frame length and magnitude response calculated. This process is repeated for all overlapped frames and the results are averaged to yield the measured magnitude room response. The measured magnitude response is then compensated with pre-determined microphone magnitude response to yield the actual magnitude response. The corrected magnitude response measured signal level is not to scale, as it has been modified by the system gain i.e. by the amp gain, microphone amp and recording gains. Also, the measured room response is invalid below 100 Hz and above 16 KHz because of measurement errors due to degraded microphone performance

2.2. Target Level and Curve

After the measurement of the room response we need to design the final room response (called as target response) using the measured room response and user preference. For example, the user preferences could be a flat target response or a bass and treble boosted target response. In any case the desired target can be represented as a flat response multiplied by the user preference. The initial aim of the equalization filter design is hence to flatten the measured response and subsequently scale the EQ filter response by the user preference.

An important factor in the design of a flat target response is the target level. The target level is the signal amplitude of all frequencies in the measured room response after EQ filtering. The target level is related to the final volume of the channel but has to be designed carefully for many reasons. Target level is the volume perceived by the listener after EQ filtering has been turned on. Theoretically there should not be a volume difference when the EQ is turned on or turned off except for the gain due to balancing the loudspeakers. In our work we designed the system to preserve signal level at the frequencies at which our ear has highest acoustic sensitivity. We calculate the target level as an average level of frequency components from 800Hz to 3 KHz. The measured room response is then divided by the target level to yield the normalized room response.

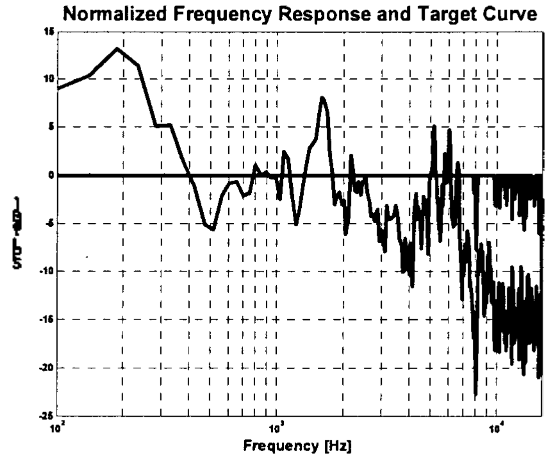

Figure 3. Normalized room response and target curve.

Due to the non minimum phase nature of the room magnitude response it is occasionally observed to have very strong zeroes. These strong zeroes will require extreme gains to correct causing issues with headroom in the system. For this purpose the limit to which any dip can be corrected during the EQ filtering is limited (Regularization). The target curve is generated using this gain limit, target level and scaled by the user preference. Using the gain limit ensures that the system needs a finite headroom. Figure 3 shows an example of a normalized room response and a calculated target curve with a 15 dB gain limiting. For the high frequency the measured response drops well below the 15dB limit and hence the high frequencies are gain limited in the target curve.

2.3. EQ Filter Design

After the measurement of the normalized room response and design of the target curve, the EQ filter design is first done by obtaining the EQ filter curve. The EQ filter curve is simply obtained by dividing the target curve with the normalized room response. Since gain limiting has already been done at the stage of the target curve generation, it is already ensured that there would be no poles occurring in the EQ filter. Subsequently the frequency sampling FIR filter design method is used to design the EQ filter. The frequency sampling method creates a filter based on a desired frequency response. Given a frequency response for the target filter, this method creates a filter whose response passes through those points. The filter design process involves generating a complex response for the EQ filter assuming the target will have a linear phase and inverse fourier transforms of the complex EQ filter response to get the filter coefficients [13]. Figure 4 shows an example of a designed EQ filter and filtered data frequency response for the Normalized room response of figure 3. The EQ filter frequency response and equalized room response have been shifted for easy comparison.

2.4. Proximity Correction

While the EQ filtering modifies the room response to the desired user shape there exists distortions in perceived sound stage because of the user proximity to some loudspeakers. The closer the listener to the speakers the louder the sound perceived. Also far speakers will have time delays associated with their sound. After the EQ filter calculations for all the channels Target level data is compared for the channels and mean Target level is calculated. The gains required for each channel to compensate for the user proximity is the difference between the mean level and channel target level.

The time difference between speakers can be calculated by first measuring the delay of each channel and then comparing the measured delay values and delaying the audio channels to synchronize with the most delayed channel. The delay calculation for each channel can be done by correlating initial MLS signal samples with the recorded MLS signal samples. The point of correlation maximization would give the channel delay.

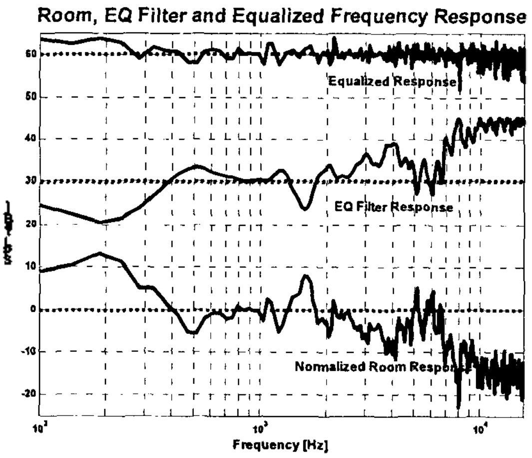

Figure 4. Room, EQ filter and equalized frequency response

2.5. Subwoofer Correction

Precise subwoofer frequency response correction would require a very high order EQ filter for appropriate frequency resolution. Down sampling and up sampling approaches can be used to improve frequency resolution but it would be computationally too expensive for implementation on our target system. Instead we implemented simple subwoofer level matching to the front speakers. Subwoofer sound level is measured using an MLS signal and gains calculated to match the level to the Target level of the front speakers. We also implemented a subwoofer phase check and correction algorithm in which the wiring for the subwoofer channel is verified by checking for the signal level in portions of overlap with the front channels. In case the subwoofer has phase inverted the response at the transition bands is degraded and this can be corrected by subwoofer signal phase inversion in software.

3. DSP IMPLEMNTATION

The target platform for the real time implementation of the algorithm described in section 2 are dual Motorola 56371 DSPs working at 180 MHz. Motorola 56371 is a 24 bit fixed point DSP targeted specifically at Audio applications with its 52 bit Accumulators, dual MAC capabilities and specialized instruction support. Since we have a total of 2 DSPs dedicated to room effects filtering available we can perform filtering up to about 2048 taps per channel at a sampling rate of 48KHz and 1024 taps at 96KHz.

The DSP implementation challenge is to be able to design the very high order FIR filter without loss of precision and in as less time as possible. Although Motorola 56371 DSP offers an extended data path for audio applications, but the filter design process of designing a 2048 order filter requires very high precision and data ranges. Hence it was decided to implement the design procedures critical to precision using floating point emulation on the DSP (24bit mantissa and 24bit exponent). Block floating point FFT's provide for very good precision and can be implemented in real time. To determine the room response we frame the data being recorded into 4096 point frames, window using a hamming window and perform a 4096 point block floating point FFT. After the FFT the current frame magnitude response is averaged by the previous frames magnitude response. The FFT and averaging calculations are done in real time during the MLS recording process. We also perform correlation to determine the system delay for the initial few frames at this stage.

At the end of the recording process we have the averaged magnitude response of the room. The data is in floating point and all subsequent calculations are done in floating point. Although floating point emulation takes a very high number of cycles but since we are working only with a 2049 points the complexity is not high. The measured room response data is compensated by the microphone response and the target level and target curves are generated from the measured data and user options followed by EQ filter response calculations and EQ filter coefficient calculations as described in Section 2. The target levels calculated for channels are used to compensate for the proximity effects along with the Subwoofer level correction. Inter channel delay and subwoofer phase correction is determined as described in section 2. Table 1 shows program and data memory for the measurement program per channel. he measurement time is the time required after the end of the recording of the MLS signal. A 4 second MLS signal is used as the excitation signal for every loudspeaker.

Program Size	21 K Words
Rom Size	1 K Words
Flash Size	8 k Words
Ram Size	46 K Words
Measurement Time	0.9 Sec

Table 1. Measurement program statistics.

The measurement stage generates EQ filter coefficients and signal gain and delay terms for proximity correction.

During the signal playback time the equalization program corrects for the room effects using the results of the measurement stage. Table 2 shows the statistics for the equalization program for 48 KHz sampling and 2048 order correction filtering. Proximity correction channel delays are limited to 20 msec.

Program Size	300 K Words
Ram Size	3 K Words
MACS	99 MACS
MIPS	50 MIPS

Table 2. Equalization program statistics.

4. EQUALIZATION PERFORMANCE

To quantify the equalization performance of our filter design method we calculated spectral deviation values before and after the equalization filtering. The spectral deviation is calculated as compared to the target level. First the deviation is calculated using the measured response and calculated target level. Next the MLS signal is inverse filtered using the calculated EQ filter and the room is excited using this signal. Then the deviation is calculated using the measured room response and earlier calculated target level. A 4096 point FFT was used to calculate the room response and it was corrected using filters with 3 different tap lengths 2048, 1024 and 512 taps. The gain limit was set as 15dB which means that extreme spectral distortions requiring correction gains beyond 15dB cannot be corrected because of head room issues. Table 3 gives details of our calculated spectral deviation results. The values are in dB. The measurement room was an acoustically treated home theater room. Results were measured for 10 points inside the room using high quality speaker system. The listening positions were all in front of the speaker system, spread around the room but away from the walls.

Location	Original Deviation	2048 Tap EQ Filter	1024 Tap EQ Filter	512 Tap EQ Filter
1	12.42	2.90	3.42	3.98
2	8.94	1.71	2.22	2.74
3	12.12	2.79	3.19	3.69
4	13.86	3.49	3.85	4.24
5	11.97	2.69	3.21	3.71
6	8.45	1.45	1.77	2.21
7	14.35	4.06	4.35	4.74
8	12.13	2.92	3.39	3.81
9	12.61	3.08	3.47	3.96
10	15.48	4.66	4.85	5.172

Table 3. Room Correction Spectral Deviation Results.

The results from table 3 demonstrate the effectiveness of our room correction method in reducing the spectral distortion caused by the room effects. It is possible that at a few points the spectral deviation is still high like location 10. This is because of the gain limit imposed because of headroom issues. In our final implementation we have used a 2048 tap filter. But for systems with lower complexity 1024 tap or 512 tap filters can also be implemented. In our measurement room reducing the EQ filter taps from 2048 to 1024 to 512 degrades the spectral deviation performance by about 0.5dB at each step.

In this paper, we described the algorithm and implementation details of a multi channel room correction system developed on a dual processor Motorola 56371 DSP system.

- Room response is measured using MLS signal playback and recording, and 2048 order correcting filters are designed.
- FIR correcting filters are designed using the frequency sampling FIR filter design method significantly reducing the spectral deviations caused by the room.
- Inter loudspeaker level and delay correction is measured and corrected to compensate for the user proximity to some speakers than others.
- Subwoofer level and phase correction algorithm is implemented.
- Floating point emulation for precision critical routines.
- Overall algorithm is designed to perform measurement in less than 1 second per channel without the use of an external PC for filter coefficient calculations.

5. REFERENCES

[1] Heinrich Kuttruff, *Room Acoustics*, Spon Press, 2000.

[2] A. G. Groh, "High-Fidelity Sound System Equalization by Analysis of Standing Waves", *Journal of Audio Engg. Soc.*, vol. 22, no. 10, pp. 795-799, Oct. 1974.

[3] J. Mourjopoulos, "Digital Equalization of Room Acoustics", *AES 92nd Convention*, Vienna, Austria, Mar. 1992.

[4] R. P. Genereux, "Adaptive Loudspeaker Systems: Correcting for the Acoustic Environment", *AES 8th In. Conf.*, Washington D.C., pp. 245-256, May 1990.

[5] O. Kirkeby, P. Nelson, H. Hamada and Felipe Orduna Bustamante, "Fast Deconvolution of Multichannel Systems using Regularization", *IEEE Transactions on Speech and Audio Processing*, vol. 6, no. 2, pp. 189-194, Mar. 1998.

[6] S. J. Elliot and P. A. Nelson, "Multiple-Point Equalization in a Room Using Adaptive Digital Filters", *Journal of Audio Engg. Soc.*, vol. 37, no. 11, pp. 899-907, Nov. 1989.

[7] S. Bharitkar and C. Kyriakakis, "A Cluster Centroid Method for Room Response Equalization at Multiple Locations", *IEEE Workshop on Applications of Signal Processing to Audio and Acoustics*, Oct. 2001.

[8] J. Leitao, G. Fernandes and A. Ferreira, "Adaptive Room Equalization in the Frequency Domain", *AES 116th Convention*, Berlin, Germany, May 2004.

[9] Yamaha Parametric Room Acoustic Optimizer (www.yamaha.com)

[10] V. Pulkki, "Virtual Sound Source Positioning using Vector Base Amplitude Panning", *Journal of Audio Engg. Soc., vol. 45, no. 6, pp. 456-466, June 1987.*

[11] Digital Room Correction System (www.tactaudio.com)

[12] M. R. Schroeder, "Integrated-impulse method measuring sound decay without impulses", *Journal of Acoustical Soc. of America*, Vol. 66, No, 2, pp. 497-500, 1979.

[13] V. K. Ingle and J. G. Proakis, *Digital Signal Processing Using Matlab*, Thompson Learning, 2001.