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THE INFLUENCE OF LOUDSPEAKER PERFORMANCE IN LOUDSPEAKER EQUALIZATION USING WIENER APPROACH

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Egalizarea răspunsului audio a unui difuzor va implica o soluție de inversare de semnal audio. Metoda LMS optimală poate îndeplini această sarcină oferind un parametru de control al compromisului dintre viteza versus performanța de egalizare. În această lucrare s-a investigat performanța filtrelor de egalizare pentru diferite răspunsuri de difuzoare simulate cu scopul de a observa, dacă există, o corelație între eroarea de filtrare, ordinul filtrului şi caracteristicile spectrale ale răspunsului difuzorului.

Equalizing a loudspeaker response will involve an audio signal inversion solution. Optimal LMS method can provide this task and offer a compromise parameter regarding equalization speed versus performance. In this paper it was investigated the performance of the equalization filters for different simulated loudspeaker responses for observing, if exists, a correlation between filter error, filter length and spectral characteristics of the loudspeaker response

Keywords: loudspeaker equalization, wiener filter.

1. Introduction

The equalization of an audio replaying chain is very important if there is a desire for true reproduction of the audio signals or if measurements in an acoustic environment are established. Analyzing the audio chain, it is observed that the electro-acoustic transducers are the main components who are responsible for the largest linear distortions but as well as non-linear distortions source. They will require a special attention in our equalization approach. These distortions can be

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observed looking into the frequency domain by the deviation from a flat spectral magnitude and from a linear phase. When referring to time domain, the distortions can be quantified by the difference between the impulse response of the loudspeaker and a Dirac delta function.

Using digital filters in equalization offers some important advantages. Digital signal processing has an ever-increasing role to play in audio systems. Practically the majority of the replaying audio systems will contain in their chain a DSP unit. Designing digital filters allows us to a smoother control of the filters parameters that will lead to a more specific correction of the measured distortions. The equalization of the audio systems can be done by a prefiltering of the stored digital signal, or during audio playback but also after, by postfiltering, if the situation allows. Different filter lengths can be used and adapting them will not require a hardware modification. Practically, for equalization we need to solve a problem of audio signal inversion, also known as impulse response deconvolution. If we denote the loudspeaker impulse response with $h(n)$, then the equalization filter described by its impulse response $w(n)$ will have to satisfy the following:

$$h(n) * w(n) = \delta(n-k) \quad (1)$$

, where "k" is a necessary delay without which, the correction filter will be non-causal.

For designing the correction filters, an often used technique is the LMS (Least Mean Squares) one offering a FIR (Finite Impulse Response) filter solution [1-3][4-8]. FIR filters are more easily to implement and optimize using less effort than for example IIR structures. It is simpler to generate a correction filter with an arbitrary magnitude and phase functions when using FIR structures. The solutions are more robust. As disadvantages, we mention that for high quality equalization, filters with larger orders are needed but the length of the filters is subject to a practical limitation due to the computation power necessary for a real time implementation. Nevertheless, we need to mention that in the last decades the digital computation power has known a spectacular grow. Even two decades ago it was possible to implement real time FIR equalization filters with orders of 200 [2] [4] with the conclusions that considerable improvements in equalization were achieved and for loudspeaker equalization, as we can see from our results, the range for the optimum filter lengths are around this value. Indeed in order to compensate for the impulse response of an acoustic systems described by a relative long impulse response with considerable non minimum phase components, very long correction filters must be used [3], but in our case the common loudspeaker responses are predominantly minimum phase.

2. Investigation Goal

In this paper it is proposed the investigation of loudspeaker equalization using FIR filters designed by a Wiener optimization method that would generate a LMS solution. We want to evaluate the dependence between the optimal filter length and an accepted (prescribed) error of the equalization process. This evaluation should be connected with the technical data of the loudspeaker.

Visual inspection of the graphically represented data is an appropriate indication of the systems performance and it is used by more than one authors [3][8]. Of course, ultimately, the best indicator will result from listening tests. The authors who used visual inspection of performances, usually observed the equalization error in time domain or frequency domain in dependency with the length of the correction filters. They estimate an appropriate length for equalization. A question to which the present study is trying to give an answer is:" Are these lengths generalizable for any loudspeaker regardless of their frequency performance or can we expect some variations." Usually in the mentioned papers, the error functions were represented only for an individual case. Using the same criteria for evaluating the equalization error, we observe the behavior of the optimal filters length when different loudspeakers with different performances are analyzed. This is useful information because one can apply directly an optimal filter correction length only by knowing the loudspeaker performance without having to measure the correction error for different lengths of filters in order to find the appropriate one, operation that can be time consuming.

3. Wiener filter theory

For inverting mixed phase type signals, an approximate of the inversion must be considered because of the non-causality problem denoted by the direct inversion of a signal and also the infinite length of the inverted signal.

Optimal filtering in the sense of LMS will represent a good solution. Consider the block diagram in figure 1 built around a linear discrete-time filter. For a thorough study of this theory, the reader can consult [9].

The filter's input consists of the loudspeaker response $h(n)$ and the filter is itself characterized by the impulse response $w(n)$. The filter produces an output $y(n)$ and this output is use to provide an estimate of a desired response denoted by $\delta(n-k)$. The estimation is accompanied by an error with characteristics of its own.

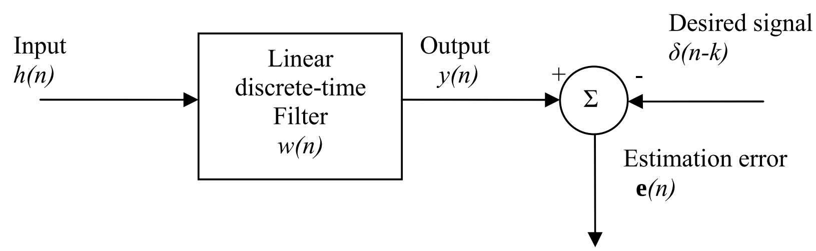

Fig. 1. Block diagram representing optimal filtering problem

In particular the estimation error *e(n)* is defined as the difference between the desired response and the filter output *y(n)*. To optimize the filter design we choose to minimize the mean square value of the estimation value. We thus define the cost function:

$$J = E[e(n)e^{*}(n)] = E[|e(n)|^{2}] \tag{2}$$

, where *E* denotes the statistical expectation operator.

We need to mention that the optimal Wiener filter is phisically realizable if the input and desired signals are the effect of a stationary random process. If we take a short glance at the loudspeaker response we can conclude that it will not be stationary on short term, but we can consider the overall loudspeaker response as the output of a stationary process.

The problem therefore is to determine the conditions for which J attains its minimum value. It can be shown that this problem of optimization comes down to solving the Wiener-Hopf Equations. If R is an M by M autocorrelation matrix of the tap inputs:

$$\mathbf{R} = E[\mathbf{h}(n)\mathbf{h}^{H}(n)] \tag{3}$$

, where $\mathbf{h}(n) = [h(0), h(1), ..., h(n-M+1)]^{T}$

, and **p** denotes the M by 1 cross correlation vector between the tap inputs of the filter and the desired response $d(n) = \delta(n-k)$:

$$\mathbf{p} = E[\mathbf{h}(n)\mathbf{d}^{*}(n)] \tag{4}$$

, then we can write the Wiener Hopf equations in their compact form:

$$\mathbf{R}\mathbf{w_0} = \mathbf{p} \tag{5}$$

Leading to

$$\mathbf{w_0} = \mathbf{R}^{-1}\mathbf{p} \tag{6}$$

, if **R** is nonsingular.

Basically we will obtain an optimal filter $\mathbf{w_0}(n)$ of order *M* which estimates in LMS sense the desired signal

$$\mathbf{w_0}(n) = [w_{o0}, w_{o1}, ..., w_{oM-1}]^T \quad (7)$$

This will be the inverse approximate of the initial signal *h(n)*, which will be in our case the loudspeaker impulse response if we impose as the desired signal the Dirac delta function $\delta(n)$. As mentioned above, we need to impose a delay in the desired signal otherwise the filter would need to be non-causal. It was observed that the equalization error was setting to a minimum for a wide range of delays. This was reported also in the previous work [1] [2] [3] [6]. Basically, if we plot the error according to the delay we can observe broad minima. For this range, the error is almost constant. The range of delays for this minima is not constant with the increase of filters length, in fact is getting larger (see Figure 2). This has a sense in a way, practically if we have a short delay, the output of the filter would have to be an impulse without the arrival to the filter of the main energy of the loudspeaker response signal, and if the delay is to long, the filter would have to output a impulse signal after a while from the moment in which the entire loudspeaker response had passed the filter. In both situations the filter is not optimized for the loudspeaker response, but as the filter's length is increasing, the time support of the filter will increase as well, making the limits for the delays larger.

One other problem is that the magnitude of the loudspeaker response is very low at low frequencies (0-100Hz) and at high frequency near the Nyquist frequency. Inverting all the response will cause a strong amplification for these spectral components which is not desired. For this we use a frequency dependent regularization factor in our cost function [7] that will introduce a new term in our optimization algorithm. As shown in [7], we can keep a good equalization for a desired frequency band and ignore in our equalization process the remaining spectral components. We will not deepen this theory, for more details see [7]. This process will change equation 6 Into equation:

$$\mathbf{w_0} = (\mathbf{R} + \beta \mathbf{R}_B)^{-1} \mathbf{p} \quad (8)$$

, where R_B is the autocorrelation matrix of a band stop filter and β is a weighting parameter that will influence how much effort will make the filter to minimize the error introduced by the new term. For example, by choosing a band stop range of 120 Hz-16 kHz and β=99 we will assure that the remaining frequency band from the spectrum will not count in the equalization process.

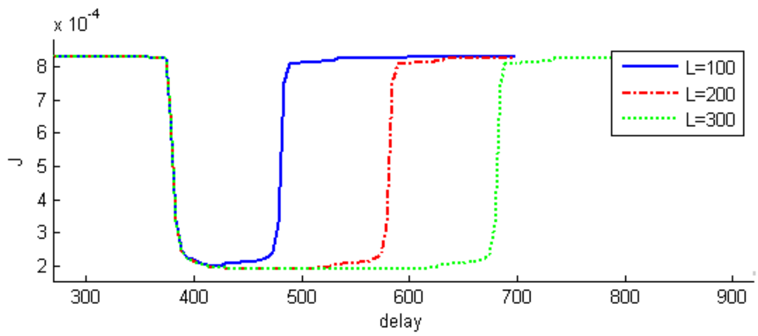

Fig. 2. Time domain equalization error calculated for different equalization filter lengths (L) for the particular Yamaha NS-40M loudspeaker response

4. Performance evaluation criteria

If our goal is to transform the loudspeaker response into a perfect impulse with the equivalent to a flat spectrum for the frequency domain, some evaluation for the performance of the inversion must be considered. The natural criteria will refer to time domain and consists of the error energy after filtering defined in equation 2. More authors [2][3][8] judged their optimal filters lengths after this criteria.

Also in frequency domain a standard deviation of the magnitude is commonly used:

$$\sigma_{\omega} = \sqrt{\frac{1}{M}\sum_{i=0}^{M-1}\left(20\cdot\log_{10}\left(\left|Y(\omega_i)\right|\right) - E_Y\right)^2} \quad (9)$$

Where $E_Y = \frac{1}{M}\sum_{i=0}^{M-1} 20\cdot\log_{10}\left(\left|Y(\omega_i)\right|\right)$ (10)

5. Experiment strategy

Because there is a lot of measurements needed for the purpose of evaluating the dependence for the performance regarding filters lengths, simulation software was developed under a Matlab platform. First of all we validated, by comparing for an individual case, our equalization calculations with a real measurement after a prefiltering equalization. The results can be viewed in Figure 3. As can be seen, very close results are obtained.

Next, three different loudspeaker responses were measured for the purpose of simulating and comparing the equalization performance. These loudspeakers are characterized by different spectral proprieties as can be observed in figure 4. If different optimal filters lengths are observed for each loudspeaker, then applying a controlled zero phase spectral distortion for one loudspeaker response, we record the behavior of the optimal equalization filter lengths for different distorsion cases. This distortion consists in band stops having as parameters the frequency band and the attenuation. Usually, this is the case in real situations; spectral falls are more common then high amplification for loudspeakers responses. We considered some limits for the distortions, so the attenuation will not exceed 20-30 dB and the stopband will not be narrower than 200 Hz. Above this limits we consider that the equalization is not proper because it will output high resonances or high amplifications, things that are very sensible to human hearing. Also the high dynamic range will limit the playback through a bad signal to noise ratio.

Conclusions of the recorded data are followed. When establishing an error versus filter's lengths plot, for a loudspeaker response, the error was calculated for every equalization filter with lengths belonging to a discrete set that covers with decreasing resolution a range of lengths between 0 and 1024. Observing the plotted data, we can notice that the error is not improving significantly for high lengths, so we decide the optimal filter length as the length that will output an energy error or a spectral deviation larger with 10% then the mean values recorded for lengths larger then 700. For each filter length situation, a step by step search for the optimum delay was calculated as well, so for every length the error would be the minimum possible.

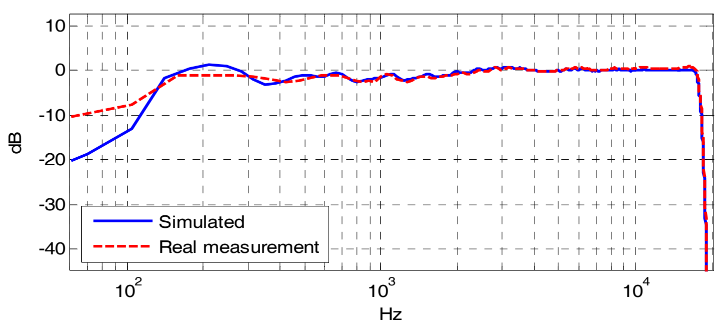

Fig. 3. Simulation versus real measurement after equalizing an M-audio BX8a loudspeaker response with a N=200 order filter

6. Results

Three loudspeakers responses were equalized for the purpose of comparing filter lengths. These were Yamaha NS-40M M-audio BX8a and a custom made loudspeaker using Mtx RTC 502 drivers. As can be seen from figure 4 and Table 1, the optimal filter lengths differ by a larger amount from one case to other. Next we took the Yamaha's response and applied some zero phase distortions. Also for each case simulated we recorded the standard deviation of the spectral magnitude of loudspeaker response. Results can be seen tables 1-7. Figure 5 presents how the distortions were applied.

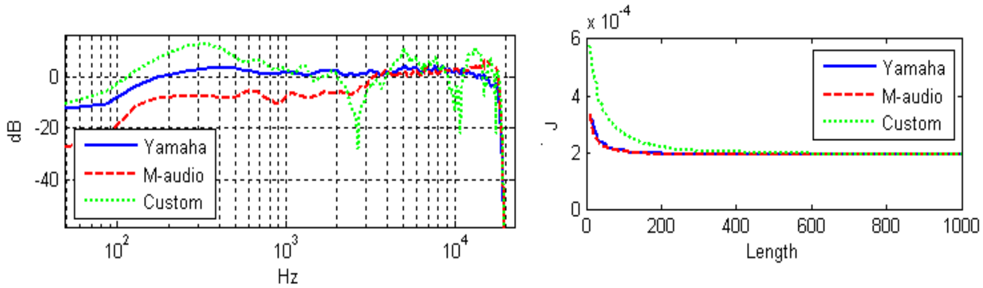

Fig. 4. a) magnitudes of three real loudspeaker responses b) their corresponding equalization error in dependence of filter's length

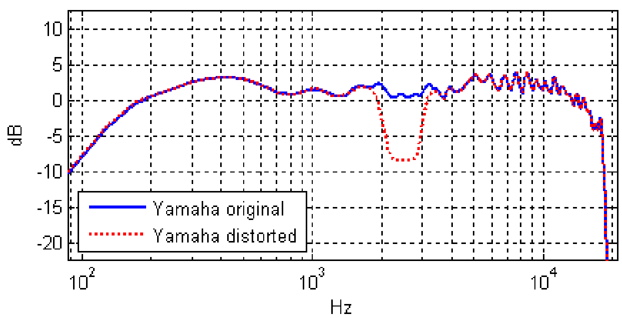

Fig. 5. Applying distortion on Yamaha NS-40M loudspeaker response

Table 1

Optimal equalization filter lengths (OEFL) calculated for three different loudspeakers

Loudspeaker type	Spectral magnitude standard deviation [dB]	Optimal filter length Based on time error
Yamaha NS-40M	1.9	67
M-Audio BX8a	3.7	68
Custom	6.8	235

Table 2

OEFL calculated for controlled distortions on 2 kHz÷2.4 kHz band for Yamaha's response

Attenuation [dB]	0	3	6	9	15	30
Spectral magnitude standard deviation [dB]	1.9	2.1	2.6	3.3	4.7	0
Optimal filter length Based on time error	67	65	66	68	78	0

Table 3

OEFL calculated for controlled distortions with constant attenuation of 20dB for Yamaha's response

Band stop [kHz]	2-2.2	2-2.5	2-3	2-4	2-6	2-8
Spectral magnitude standard deviation [dB]	2.7	3.6	4.8	6.3	7.6	8.3
Optimal filter length Based on time error	75	80	79	68	67	65

Table 4

OEFL calculated for controlled distortions with constant two band stops attenuation of 15dB each for Yamaha's response

Band stop [kHz]	2-2.2 / 10-10.2	2-2.5 / 10-10.5	2-3 / 10-11	2-4/ 10-12	2-5 / 10-13
Spectral magnitude standard deviation [dB]	2.6	3.6	4.7	5.9	6.7
Optimal filter length Based on time error	97	133	109	103	98

Table 5

OEFL calculated for controlled distortions with constant two band stops attenuation of 30dB each for Yamaha's response

Band stop [kHz]	2-2.2 / 10-10.2	2-2.5 / 10-10.5	2-3 / 10-11	2-4/ 10-12	2-5 / 10-13
Optimal filter length Based on time error	109	279	163	103	93

Table 6

OEFL calculated for controlled distortions with constant three band stops attenuation of 15dB each for Yamaha's response

Band stop [kHz]	2-2.2 / 6-6.2 / 10-10.2	2-2.5 / 6-6.5 / 10-10.5	2-3 / 6-7 / 10-11	2-4/ 6-8 / 10-12
Optimal filter length Based on time error	106	156	122	108

7. Conclusions

Starting from the difference on optimal equalization filter length for three real situations of loudspeaker responses, correlation of this parameter with the spectral magnitude proprieties of the loudspeakers were investigated by simulations. Based on these results we can state that the global spectral cues like standard deviation or maximum dynamic range are not correlated with the optimal filter length.

Analyzing the spectral falls from the response, we can observe that if the falls are narrower, in every situation listed, the optimal length is increasing, with exceptions for spectral minima thinner than 500 Hz where it seems that the optimal length has a decrease. Probably small bands like this don't have a noticeable influence on global errors.

The attenuation of the falls are also correlated with the filters lengths, a higher attenuation involves a necessary of more filter coefficients.

Also with the number of falls in the spectral magnitude rising, the optimal filters length is increasing. When the spectral minima is wide, the attenuation doesn't have a noticeable influence, but as the fall is becoming thinner, the attenuation has a major role (see table 2, 4 and 5). This is the biggest perceivable difference noticed.

Analyzing the frequency criteria, the same trend was noticed for all statements. With all these distortions, the optimal length is less than 300 which is more than enough for today's computation speed for real time implementation. Practically, if the loudspeaker response doesn't have a big dynamic range (no more than 30 dB) for our band of interest, a 300 length filter can do an optimal equalization.

After grouping all the results, the major conclusion is that for normal loudspeakers with a magnitude dynamic range under 30 dB, a 300 order LMS filter will do the job in almost every case. Different correlation were noticed and discussed.

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