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# DESIGN AND IMPLEMENTATION OF HIGH-ORDER DIGITAL EQUALIZERS FOR AUDIO SIGNAL USING MATLAB AND DSK TMS320C6711

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**Abstract.** In this paper, the transfer function of the seventh order digital graphic equalizer is calculated. The gain responses of the digital filters, of individual equalizers and of overall graphic equalizer are designed by MATLAB and implemented by DSK TMS320C6711. These gains can be controlled independently by adjusting the parameters C, $\alpha$ and $\beta$ of each section in the digital graphic equalizer.

## 1. Introduction

In the audio and musical instruments , the equalizers are used to enhance performance of the transmitter channels or to improve the quality of sound reaching to the listeners. A typical equalizer consists of a low frequency shelving filter and three or more peaking filters with adjustable parameters to provide adjustment of the overall equalizer frequency response over a broad range of frequencies in the audio spectrum. In a parametric equalizer, each individual parameter can be varied independently without effecting the parameters of the other filter blocks in the equalizer. In a graphic equalizer , its consists of a cascade of peaking filters with fixed center frequencies but adjustable gain levels.

The major applications of equalizers are to correct and to improve certain types of problems that may have occurred during the processing or the transfer process and to alter or to reduce the noise. The adaptive equalizers are basically an adaptive filter FIR with coefficients that are adjusted by the LMS algorithm to compensate channel distortions caused by intersymbol interferences (ISI).

In this paper, allpass filters are employed to design and to realize high order equalizers for audio and musical signals. The purpose of these equalizers is to increase the desired frequency components and to reduce the undesired frequency components in the sound range by modifying the gain response.

## 2. Structure of high order equalizer

A high order equalizer is created by connecting a cascade of one first-order with one or more second-order equalizers . The frequency response of overall equalizer can be controlled by adjusting the center frequencies of each section in the cascade. Figure (1) shows the block schema of a cascade of a seventh equalizer which consist of one first-order and three second-order equalizers. In this block schema, $A_1(z)$ is transfer function of the first-order allpass filter, while $A_2(z)$, $A_3(z)$, $A_4(z)$ are transfer functions of the second-order allpass filters.

The first-order equalizer is created by adding one low-pass filter and one high-pass filter with the multiplier coefficients $C_1/2$ and $1/2$, respectively and is characterized by the following transfer function

$$H_1(z) = \frac{Y_1(z)}{X(z)} = \frac{C_1}{2}\left[1 - A_1(z)\right] + \frac{1}{2}\left[1 + A_1(z)\right], \tag{1}$$

where $C_1$ is a positive parameter; $A_1(z)$ is a first-order allpass transfer function given by

$$A_1(z) = \frac{\alpha_1 - z^{-1}}{1 - \alpha_1 z^{-1}}$$

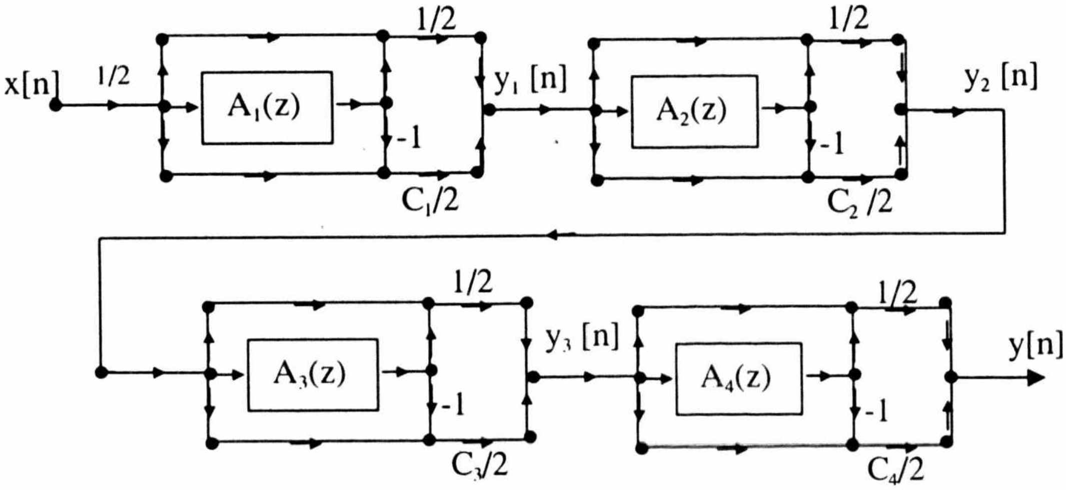


***Figure 1.*** The block schema of a seventh - order graphic equalizer.

The frequency response of this first section $H_1(z)$ can be varied by varying the values of parameters $C_1$ and $\alpha_1$. Parameter C controls the amount of boost or cut at low frequencies, while the constant $\alpha_1$ controls the boost or cut bandwidth.

The corresponding input-output relation of the first-order equalizer is described by following difference equation

$$y_1[n] = \frac{1}{2}\,[(C_1+1) + (1-C_1)\alpha_1]x[n] + \frac{1}{2}\,[(C_1-1) - (C_1+1)\alpha_1]x[n-1] + \alpha_1 y_1[n-1]\cdot \tag{3}$$

That shows clearly that the coefficients of difference equation can be adjusted by varying the parameters $C_1$ and $\alpha_1$.

The transfer function of the $i^{th}$ second-order equalizer is given by

$$H_i(z) = \frac{C_i}{2}\left[1 - A_i(z)\right] + \frac{1}{2}\left[1 + A_i(z)\right],\ i = 2, 3, 4, \tag{4}$$

where

$$A_i(z) = \frac{\alpha_i - \beta_i(1+\alpha_i)z^{-1} + z^{-2}}{1-\beta_i(1+\alpha_i z^{-1} + \alpha_i z^{-2}}, \; i = 2, 3, 4 \quad (5)$$

The relations (4) and (5) show that the $i^{th}$ equalizer is created by combining one bandpass filter with one bandstop filter. The center frequency and the 3-dB bandwidth of each filter can be varied by varying the values of parameters k, $\alpha_i$ and $\beta_i$. These parameters of each equalizer can be tuned independently without effecting the parameters of the other sections . Therefore the frequency and magnitude response of the overall equalizer can be controlled by adjusting these parameters.

The center frequency $\omega_{0i}$ is controlled by the parameter $\beta_i$, because which is determined by the following relation

$$\omega_{0i} = \arccos(\beta_i) \cdot \quad (6)$$

The parameter $\alpha_i$ determines the 3-dB bandwidth $B_{wi}$ of each equalizer by relation (7)

$$B_{wi} = \arccos\left(\frac{2\alpha_i}{1+\alpha_i^2}\right) \cdot \quad (7)$$

The magnitude response of the $i^{th}$ equalizer is controlled by parameter $C_i = H_i(e^{j\omega_0})$.

The transfer function of the overall equalizer as on figure(1) given by

$$H(z) = \frac{Y(z)}{X(z)} = H_1(z)H_2(z)H_3(z)H_4(z) \cdot \quad (8)$$

## 3. Numerical results

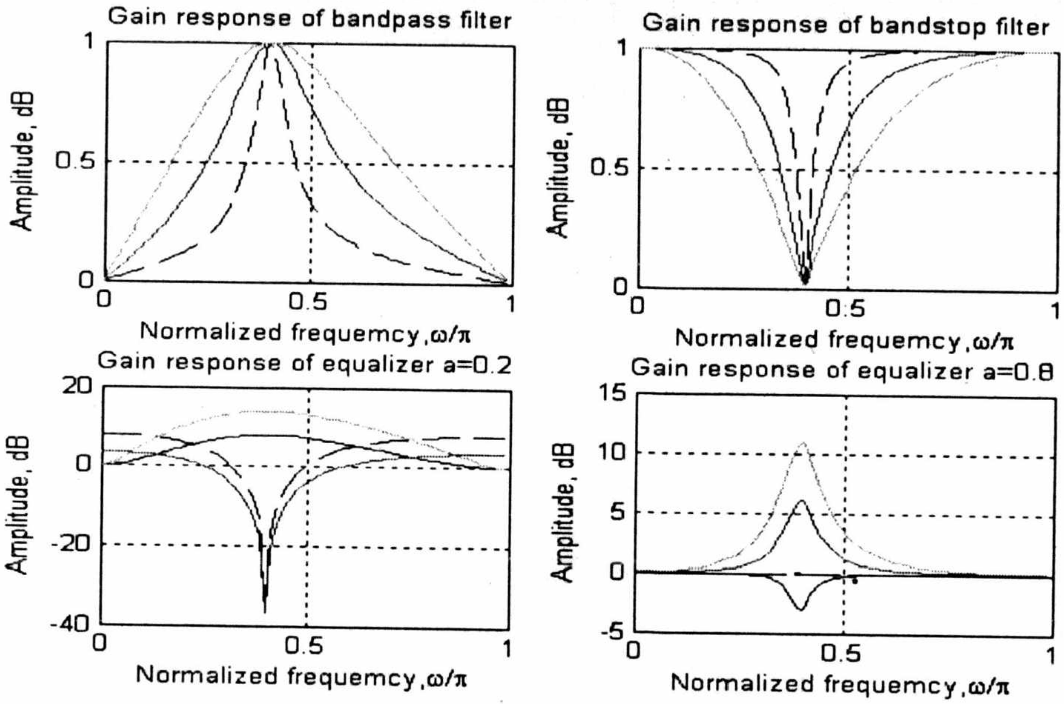


*Figure 2.* Gain response of the bandpass, bandstop and second-order equalizers with the different values of C, $\alpha$ and $\beta$.

A second-order equalizer is built by adding one bandpass filter with one bandstop filter. Figure2 shows the gain responses of these filters and of the equalizer simulated with different parameters C, $\alpha$ and $\beta$. The bandpass and bandstop filters are designed with the values of $\alpha$ : $\alpha_1$ =0.8; $\alpha_2$ =0.5; $\alpha_3$ =0.2 and $\beta$ = 0.315. These filters are employed for implementing two second-order equalizers; the first equalizer with the parameters : $C_{10}$ =1.5; $C_{20}$ =2.5; $C_{30}$ =5; $C_{40}$ =0.5; $\alpha_3$ =0.2; $\beta$=0.8 and $C_1$ =1; $C_2$ =2; $C_3$ =3.5; $C_4$ =0.7; $\alpha_1$ =0.8; $\beta$ = 0.315 .

By connecting in cascade of one first-order equalizer with the second-order equalizers , we can built the higher- order graphic equalizers as plotted on the figure1. The figure 3 and 4 plot the gain responses of the bandpass, bandstop filters, equalizers and seventh-order graphic equalizer obtained by synthesizing these filters and individual equalizers from equations (4) and (8). Figure3 is plot of gain responses with the parameters of values: $\alpha$ = 0.1584; $\beta_2$ = 0.809; $\beta_3$ = 0.309; $\beta_4$ = - 0.809 and $C_1$ = 1.3; $C_2$ =1.2; $C_3$ = 0.95; $C_4$ =1.1 and figure4 with $\alpha$ = 0.7267; $\beta_2$ = 0.7071; $\beta_3$ = 0.1564; $\beta_4$ = - 0.7071; and $C_1$=1.3; $C_2$=2.75; $C_3$=3.65; $C_4$ = 3.21. Figure 5 is the impulse response of graphic equalizer which has frequency response given on the figure 4.

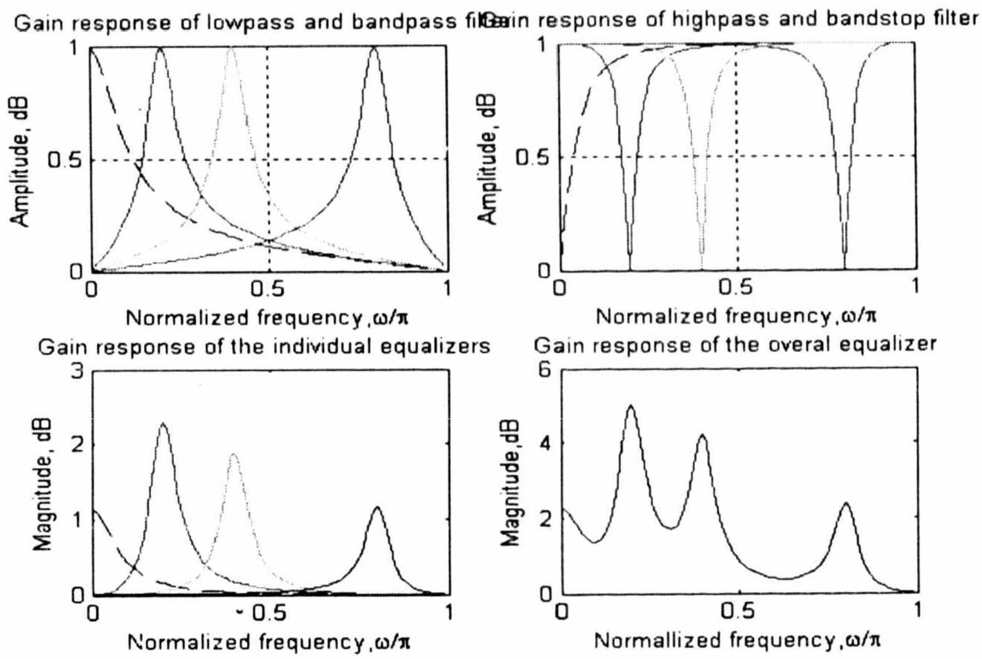


*Figure 3.* Gain response of the bandpass, bandstop filters, individual equalizers and seventh-order graphic equalizer with: $\alpha$ = 0.8; $\beta_2$ = 0.809; $\beta_3$ = 0.309; $\beta_4$ = - 0.809; and C1 = 1.3; C2 = 1.7; C3 = 1.55; C4=1.31.

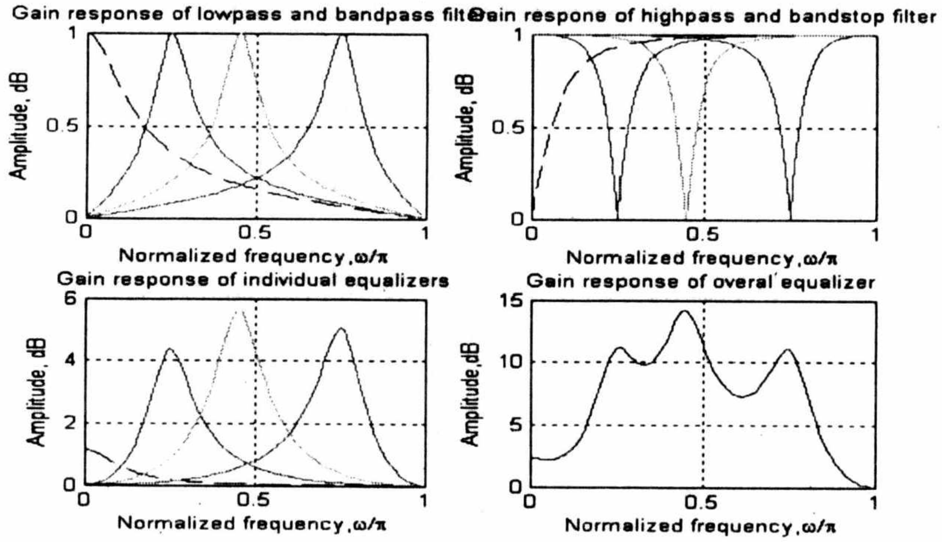


*Figure 4.* Gain response of the bandpass, bandstop filters, individual equalizers and seventh-order graphic equalizer with: α=0.7267; β2=0.7071; β3=0.1564; β4= - 0.7071; and $C_1$=1.3;$C_2$=2.75; $C_3$ =3.65;$C_4$ =3.21.

The plots show that the gain response of each equalizer and can be regulated independently without effecting the parameters of the other equalizers and hence the gain response of overall equalizer can be controlled by regulating the parameters of each individual equalizer. Therefore, the desired frequency components can be increased or reduced by regulating the parameters C, α or β, respectively.

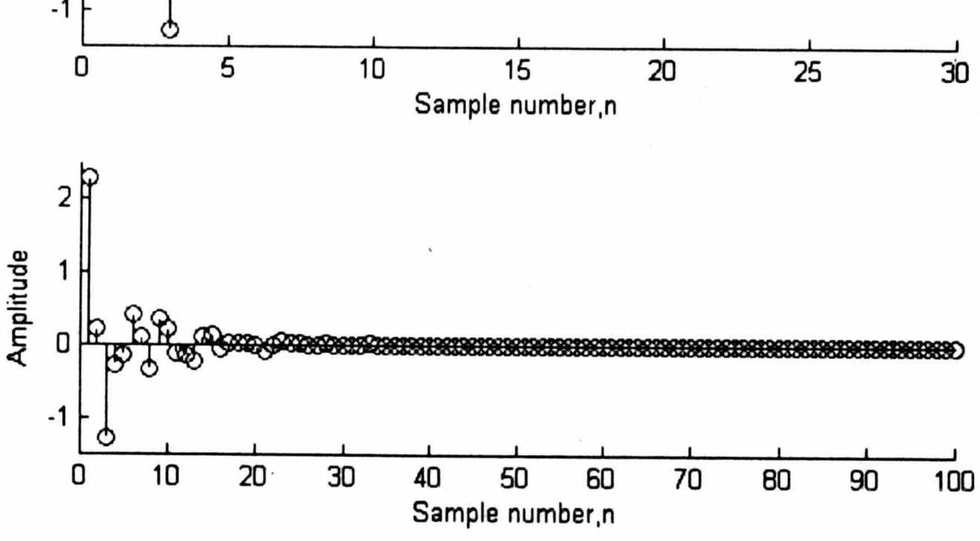


*Figure 5.* Impulse response of the graphic equalizer with frequency response given on the figure 4.

The coefficients of overall graphic equalizer are printed in the following table

**coeff.h** =

{
2.2877; 0.2110; -1.2788; -0.2758; -0.1347; 0.4113; 0.1148; -0.3444; 0.3659; 0.2325; -0.1208; -0.1567; -0.2313; 0.1193; 0.1500; -0.0649; 0.0196; 0.0172; 0.0234; 0.0031; -0.0916; -0.0076; 0.0445; 0.0136; 0.0105; -0.0173; -0.0002; 0.0143; -0.0162; -0.0092; 0.0069; -0.0021; -0.0059; 0.0012; 0.0017; 0.0037; -0.0006; -0.0037; 0.0007; 0.0005; -0.0003; -0.0002; -0.0006; 0.0012; 0.0006; -0.0008; -0.0003; -0.0001; 0.0003; 0.0002; -0.0003; 0.0001; 0.0002; -0.0000; -0.0001; -0.0002; 0.0000; 0.0001; -0.0000; ...0.0000}.

## 4. Implementation of a high - order equalizer using DSK TMS320C6711

The above seventh-order graphic equalizer can be implemented by employing DSK TMS320C6711. In this instrument, the four sets of coefficients of graphic equalizer designed by MATLAB in the above table is contained in the file graphicEQcoeff.h. Both the input samples and the set of coefficients are transformed into the frequency domain. Because the filtering is implemented by fast convolution with overlap-add method. The complex FFT and IFFT are carried out on the floating point DSK TMS320C6711.

The program graphicEQ.C which implements this seventh-order equalizer is tested using an input voice file *Theforce.wav* added a sinusoid of the frequency 950Hz which is generated by bass frequency generator. In the output of overall equalizer, this sinusoidal signal is attenuated, because the dip of the gain response of equalizer occurs at this frequency component. The slider file *graphicEQ.gel* allows to control four frequency bands of overall equalizer independently. The input, output signals and their spectrum of the overall equalizer can be obtained with a digital oscilloscope, with a signal analyzer, with the CCS-window or with an earphone.

## 5. Conclusion

By using the first-order and second-order allpass filters , the lowpass, highpass, bandpass and bandstop filters are built. These filters are the basic components to constitute the individual equalizers. Therefore the overall graphic equalizer has a very simple structure. That means that the implementing the FIR filter is carried out rapidly not only on the software but also on the hardware. Because, it allows to reduce a great number of computations as well as the number of delays, adders and the coefficient multipliers. The MATLAB and DSK TMS320C6711 programs permit to control flexibly the parameters of each individual equalizer and hence the gain response of the overall graphic equalizer can be controlled flexibly in a desired range of frequency.

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