FIR and IIR Filters for Sound Equalization Systems

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Received: November 2019 Revised: January 2020 Accepted: March 2020

ABSTRACT:
IIR and FIR filters are widely used in digital signal processing. The choice of one of these methods depends on the design requirements. In this paper, the two filter design methods are compared in audio equalizer system design. Audio equalization systems are used in electric guitars, pianos, and other electric instruments. It is also an integral part of audio players, audio amplifiers and sound recording systems. The design includes a six-band amplifier that can be used to modify the signal amplitude over the entire audio frequency spectral range. Digital filters were transformed from Chebyshev analog prototypes using Bilinear Transformation Procedures on IIR filters. The Window method was used to design the FIR filter with the same specifications as the IIR filter. The results show that IIR and FIR can be used to design equalizer systems. But one has advantages over the other. Filters can be designed with low order IIR methods, but the FIR requires high order filter designs. On the other hand, in the FIR filter design method, the linearity of the phase response is much better.

KEYWORDS: IIR filter, FIR Filter, Audio Equalizer, Bilinear Method, Window Method.

1. INTRODUCTION

Filters play a major role in digital signal processing, where IIR and FIR filter design methods are the most widely used. IIR filter stand for infinite impulse response filter and the output depends on the current and past inputs as well as past outputs. FIR filter is the abbreviation for finite impulse response and its output depends on current and past inputs. In this paper the two filters are used to stablish an audio equalizer. Using the audio system, both types of filters can be considered from zero to folding frequencies using low pass, band pass and high pass filters.

There are several methods to implement these two types of filters. IIR filters can be designed using bilinear transformation, Impulse Invariant and pole-zero placement design techniques. FIR filter types can be performed using, frequency sampling, window method and optimal design method. Window design method is most common for FIR filter design and bilinear transformation technique has most application in IIR filter type design. In this paper, window design method and bilinear transformation design procedure were used for FIR filter and IIR filter design, respectively.

There has been many research that shows IIR and FIR filters applications. From the previous researches can be seen that finite bit precision affects both FIR and IIR filter types, however, IIR filter is more sensitive to finite bit precision effect [1]. FIR filters are used when the linear characteristics of the filter are important in systems where the IIR filter is less linear than the FIR filter. The linearity characteristic of an FIR filter is the result of the linear phase response of the filter [2]. Designing FIR filters requires higher order tapping compared to IIR filters. High-order tapping is problematic for some applications, so IIR filters are preferred for those applications [3], [4]. On the other hand, FIR filters are more stable than IIR filters and do not require feedback, so FIR filters are preferred for most applications [5], [6].

Next, the filter specifications of the six-band equalizer are presented [7]. The filter is then designed using the IIR and FIR filter design methods. A sample audio signals was provided to confirm the results for both design methods. Eventually the results will be discussed.

2. METHODOLOGY

The equalizer system design includes a six-band filter design for the entire audio signal frequency spectrum. This property gives the user the ability to modify the amplitude of each isolated frequency band of the signal or audio as desired. The target audio of the system is a signal with a sampling frequency of 44100 Hz, which can support the entire audio frequency range. The filters listed in the first column of the following table show the filter types and specifications required for the audio equalizer system. In Table 1, f_{sl} , f_{pl} , f_{ph} and f_{sh} are low stop, low pass, high pass and high stop frequency edges, respectively.

Table 1. Audio equalizer system filter specification.

Filter No.	Filter type	f_{sl}	f_{pl}	f_{ph}	f_{sh}
1	Low-pass	1100 Hz	700 Hz	-	-
2	Band-pass	400 Hz	700 Hz	1700 Hz	2000 Hz
3	Band-pass	1400 Hz	1700 Hz	2700 Hz	3000 Hz
4	Band-pass	2300 Hz	2700 Hz	3900 Hz	4300 Hz
5	Band-pass	3500 Hz	3900 Hz	5300 Hz	5700 Hz
6	High-pass	-	-	5300 Hz	4700 Hz
Sampling frequency: 44100 Hz, Passband ripple : 1 dB, Stopband attenuation : 40 dB					

In audible signals, low frequencies have a greater magnitude and mostly consist of the main information of the audio. Therefore, the filter design mainly focuses on frequencies below 5300 Hz. The frequency spectrum is divided into six sections, including a low pass filter, four band pass filters, and the last one is a high pass filter. The high pass filter separates frequencies above 5300 Hz. Therefore, the user can adjust any of the six-band frequency responses in Table 1. The ripple of the pass band was selected at 1 dB for all filters and the minimum attenuation of the stop band was set at 40 dB. The following sections describe the filter design using the IIR and FIR methods, respectively.

2.1. Audio Equalizer Design using IIR Filters

The Bilinear Transformation design procedure was implemented to design filters using the IIR method. In this procedure, the digital filter specifications are prewarped to the analog frequency characteristics. The analog filter prototype transformation was then used to design the analog filter equivalent to the desired digital filter. Finally, using the bilinear transformation, the analog filter becomes a digital filter [8]. Table 2 shows the prewarped analog frequencies equivalent to the digital frequencies in Table 1 using equation 1.

$$\omega_a = 2 f_s \tan(\pi \frac{f_d}{f_s}) \tag{1}$$

Where ω_a is the prewarped analog frequency in rad/s, f_d is the digital frequency edge and f_s is the sampling frequency. In Table 2, ω_{asl}, ω_{apl}, ω_{aph} and ω_{ash} are analogue prewarped low stop, low pass, high pas and high stop frequency edges in $rad/\sec$, respectively. Also, $\omega_0 = (\omega_{apl}.\omega_{aph})^{0.5}$ is the center frequency and $W = \omega_{apl} - \omega_{aph}$ is the filter bandwidth.

Table 2. Prewarped analogue frequencies of digital filters.

Filter No.	ω_{asl}	ω_{apl}	ω_{aph}	ω_{ash}	ω_0	W
1	6926 rad/s	4402 rad/s	-	-	-	-
2	2514 rad/s	4402 rad/s	10734 rad/s	12652 rad/s	6874 rad/s	6294 rad/s
3	8826 rad/s	10734 rad/s	17177 rad/s	19142 rad/s	13579 rad/s	6294 rad/s
4	14582 rad/s	17177 rad/s	25155 rad/s	27896 rad/s	20787 rad/s	7558 rad/s
5	22458 rad/s	25155 rad/s	34979 rad/s	37922 rad/s	29663 rad/s	8826 rad/s
6	-	-	34979 rad/s	30686 rad/s	-	-

Next, a prototype transformation is performed using the low-pass prototype to design analog low-pass, band-pass, and high-pass filters. In this study, Chebyshev low-pass prototype transfer with 1dB ripple was selected due to sharp band shift. Prototype order can be calculated using the following equation [9]:

$$n \geq \frac{\cosh^{-1}\left(\sqrt{\frac{10^{0.1A_s}-1}{10^{0.1A_p}-1}}\right)}{\cosh^{-1}(\nu_s)} \tag{2}$$

In this equation, A_s is the stopband attenuation (dB) at normalized stopband frequency edge ν_s and A_p is the passband ripple (dB) at normalized passband frequency 1. For low-pass filter $\nu_s = \omega_{as} / \omega_{ap}$, for band-pass filter $\nu_s = (\omega_{ash} - \omega_{asl}) / (\omega_{aph} - \omega_{apl})$ and for high-pass filter $\nu_s = \omega_{ap} / \omega_{al}$. In Table 3, the prototype order is calculated for each filter using equation 2.

Table 3. Calculated prototype order number for each filter.

Filter No.	1	2	3	4	5	6
Filter Type	Low pass	Band Pass	Band pass	Band pass	Band pass	High pass
n	5.83	5.64	5.7	5.43	5.82	11.42

Table 3 shows the calculated prototype minimum orders. Rounding the order value, 6 becomes the low-pass and band-pass filter order, and 12 becomes the high-pass filter order. An order 6 Chebyshev low-pass prototype with 1dB ripple in the passband is[10]:

$$H_{p6}(s) = \frac{0.0614}{s^2 + 0.1244s + 0.9907} \times \frac{1}{s^2 + 0.3398s + 0.5577} \times \frac{1}{s^2 + 0.4641s + 0.1247} \quad (3)$$

The Chebyshev low pass prototype of order 12 can be obtained by cascading two $H_{p6}(s)$ in Equation 3, as shown in Equation 4.

$$H_{p12}(s) = H_{p6}(s) \times H_{p6}(s) \quad (4)$$

Next, to get a transition from a low pass prototype to a low pass filter, s must be replaced by s/ω_{apl}. To convert from the low pass prototype to the band pass filter, s must be replaced by $(s^2+\omega_0{}^2)/sW$. In order to convert the low-pass prototype into a high-pass filter, s must be replaced by ω_{aph}/s. The transformation process can be performed using the specified Matlab functions.

Finally, to obtain the digital filter transfer function $H(z)$ from the designed analog filter transfer function $H(s)$, the bilinear transformation is performed where s is replaced by $2(z-1)/T(z+1)$. The frequency spectrum of IIR filters designed in accordance with the specifications given in Table 1, are shown in the following figure. As shown in Figure 1, the six filters are designed to cover the entire frequency spectrum, which allows modifying the magnitude of the signal for any desired frequency.

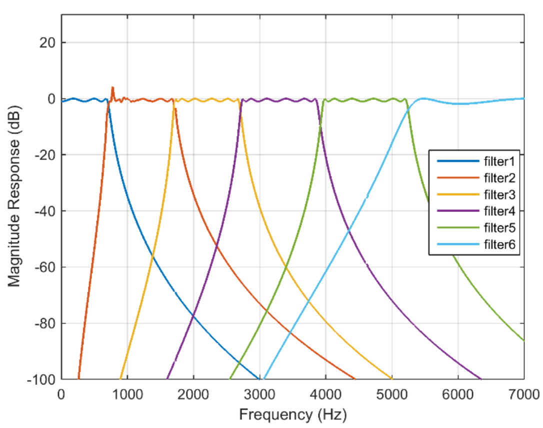

Fig. 1. Six designed IIR filters for the entire audio frequency spectrum.

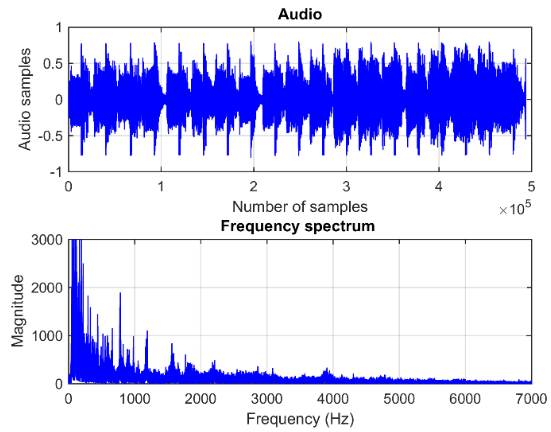

Fig. 2. A sample audio signal and its frequency spectrum.

The resulting transfer function of the digital filters, shown in Figure 1, can be tested using a sample audio signal. The selected sample signal has 500k samples at 44100 sampling frequencies. Figure 2 shows a sample signal and its frequency spectrum. By applying a sample signal to the designed audio equalizer shown in Figure 1, the frequency spectrum of the signal will be divided into six parts, as shown in Fig. 3.

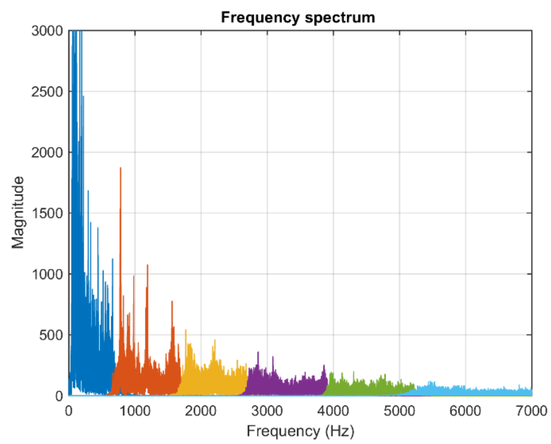

Fig. 3. Divided frequency spectrum using audio equalizer into 6 parts.

2.2. Audio Equalizer Design using FIR Filters

To design an audio equalizer using the FIR filter, a Fourier transform design with a window function called the window method was used. The window method needs to calculate the impulse response $h(n)$ [11]. And the impulse response is limited to the sequence of the main samples $2M+1$. Next, the filter coefficient b_n can be found as $b_n = h(n-M)$. The following relationships are used to calculate the idealized truncated $h(n)$ for standard low-pass, band-pass and high-pass filters[12]. For the low-pass filter ideal $h(n)$ would be as:

$$h(n)=\begin{cases}\dfrac{\Omega_c}{\pi} & n=0\\[2ex] \dfrac{\sin(\Omega_c n)}{n\pi} & n\neq 0,-M\le n\le M\end{cases} \quad (5)$$

For the high-pass filter the equation will be as:

$$h(n)=\begin{cases}\dfrac{\pi-\Omega_c}{\pi} & n=0\\[2ex] -\dfrac{\sin(\Omega_c n)}{n\pi} & n\neq 0,-M\le n\le M\end{cases} \quad (6)$$

The equation 7 shows the idealized truncated $h(n)$ for band the pass filter.

$$h(n)=\begin{cases}\dfrac{\Omega_h-\Omega_l}{\pi} & n=0\\[2ex] \dfrac{\sin(\Omega_h n)}{n\pi}-\dfrac{\sin(\Omega_l n)}{n\pi} & n\neq 0,-M\le n\le M\end{cases} \quad (7)$$

Where Ω is the normalized frequency and can be given as $\Omega = 2\pi f / f_s$ .The window function weighs the impulse response of the filter to minimize the cutting effect on the filter. There are certain types of window functions, such as Rectangular, Hanning, Hamming and Blackman windows, which can be used in the FIR filter design. The selection of the appropriate window depends on the properties of the filter. In this study, the Hanning window is chosen because the ripple of the passband is 1 dB and the attenuation of the stop band is 40 dB in the filters. In the Hanning filter, the order of filter N is $N = 3.1/\Delta f$ [13] and $\Delta f = \left| f_{as} - f_{ap} \right| / f_s$. The cutoff frequency f_c is also specified as $f_c = (f_{ap} + f_{as})/2$. Table 4 provides the design specifications for each filter in Table 1.

Table 4. FIR filters specifications.

Filter No.	Filter type	Filter order N	Low cutoff frequency Ω_{cl}	High cutoff frequency Ω_{ch}
1	Low-pass	343	0.0408π rad	-
2	Band-pass	457	0.0249π rad	0.0839π rad
3	Band-pass	457	0.0703π rad	0.1293π rad
4	Band-pass	343	0.1134π rad	0.1859π rad
5	Band-pass	343	0.1678π rad	0.2494π rad
6	High-pass	229	-	0.2268π rad

Matlab program can be used to get the coefficients of the FIR filters listed in Table 4. Figure 4 shows the result filter.

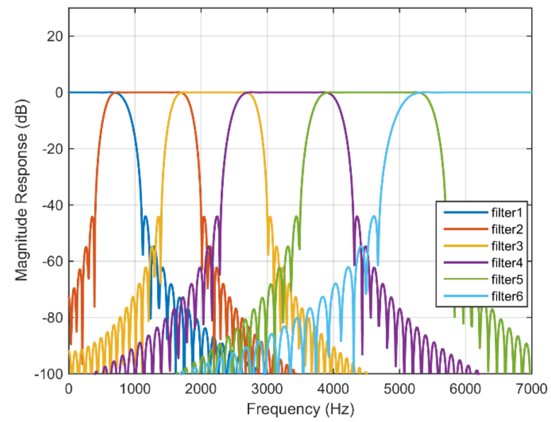

Fig. 4. The FIR filters for the audio equalizer system.

If the equalizer designed using the FIR filters is implemented on the sample audio signal, the resulting frequency spectrum would be as shown in Figure 3. the magnitude of the desired signal in the spectrum can easily be modified using the equalizer. Figure 5 shows a sample audio spectrum with the frequency range 2700-3900 Hz amplified.

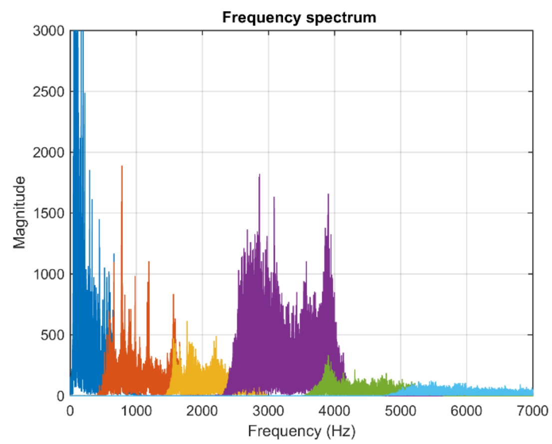

Fig. 5. Modified audio spectrum using the equalizer system.

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2.3. Phase Response

The phase response for the IIR filter and the FIR filter was calculated using Matlab. Figures 6 and 7 show the frequency and phase response of a bandpass filter at 2700 t0 3900 Hz using IIR and FIR filters, respectively. As expected, the FIR filter had more linearity in its phase response compared to the IIR filter.

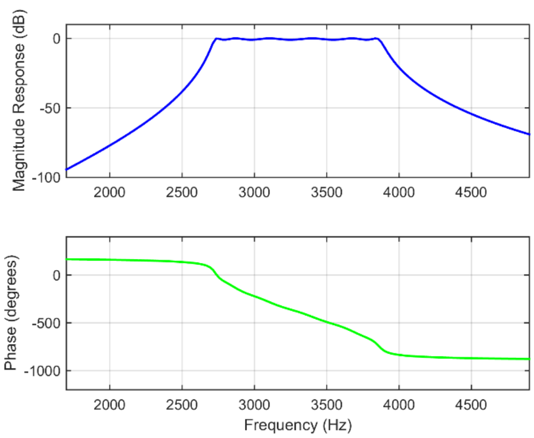

Fig. 6. FIR frequency response from 2700 to 3900 Hz.

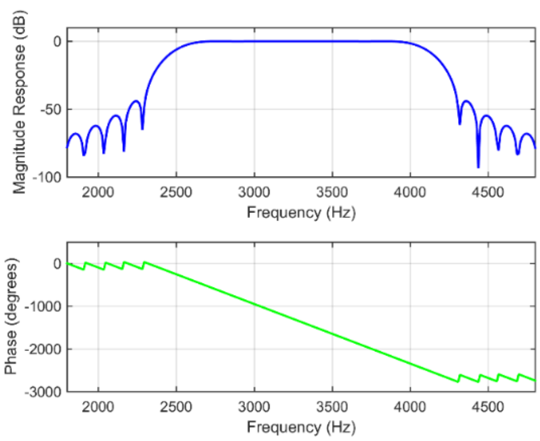

Fig. 7. FIR frequency response from 2700 to 3900 Hz

3. CONCLUSION

In this study, two digital FIR and IIR filter designs were used to design similar equalizers with known bandwidth specifications. In the equalizer system design, the audio frequency spectrum is divided into six parts, which include a low pass filter, four bandpass filters and a high pass filter. For IIR filters, the Bilinear Transformation design procedure was implemented, and the FIR filter design used the window method. For design, both FIR and IIR filters are suitable. However, one has certain advantages over the other. IIR filter designs have fewer filter orders than FIR filters, which require less math and computation time. In contrast, FIR filters have a higher linearity in phase response than IIR filters. The passband of the FIR filter had zero ripple, but the IIR filter showed less ripple in the stopband. Choosing one filter type from another is a matter of design priority. If the phase linearity and flat pass band are priorities, the FIR filter is selected; otherwise, the IIR filter can be used.

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