DESIGN AND PERFORMANCE ANALYSIS OF BAND PASS IIR FILTER FOR SONAR APPLICATION

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ABSTRACT

In recent years due to the magnificent development of filter designs, designers took attention in this research area. Filters are used to separate or combine different frequencies. In this paper IIR filter has been designed and simulated using different techniques. Butterworth, Chebyshev1, Chebyshev2 and Elliptic filters are the basic prototype filter from which we designed IIR filter. The performance of all the developed filters has been analyzed and compared by using different filter order in MATLAB. It can be observed from the simulated result that the performance of Butterworth filter is better at lower order and Chebyshev2 and Elliptic filter shows better performance with increase in filter order in SONAR frequency range.

Key Words: *Band pass, Butterworth, Chebyshev1, Chebyshev2, Elliptic, IIR, SONAR.*

1. INTRODUCTION

ADSP is the process of mathematical analysis of a signal to modify and improve its characteristics. Digital signal processing (DSP) refers to various techniques for improving the accuracy and reliability of digital communications. The theory behind DSP is quite complex. Basically, DSP works by clarifying the levels or states of a digital signal. ADSP circuit is able to differentiate between human-made signals, which are orderly, and noise, which is inherently chaotic [1]. The working of digital signal processor is as follows:

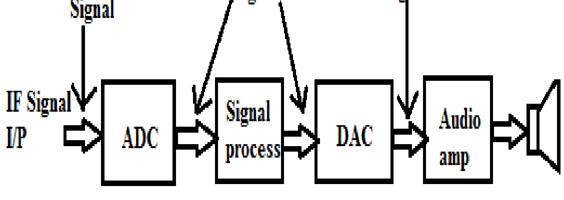

Fig. 1 Digital signal processing

The digital signal processing is shown in Fig. 1 where we have to convert the analog signal into digital signal by using A/D converter and then the function of digital processor starts. After transmission of digital signal successfully at the receiver terminal we need D/A converter to convert the digital signal into its analogous form. The basic task of DSP is to transform, filter and compress the analog signal. After doing the task we are able to store information signal more efficiently. The advantages to using DSP techniques are Reproducibility, Programmability, Stability and High reliability etc. The applications of DSP techniques are Audio signal processing and compression, Digital image processing video compression, Speech processing and recognition, RADAR, SONAR Financial signal processing, Biomedicine etc.

The basic task of DSP is filtering. Filtering is the one of the most powerful and important tool of DSP. Digital filters are capable of performing that specifications which are extremely difficult, to achieve with an analog implementation. In addition, the characteristics of a digital filter can be easily changed under software control. Many digital systems use signal filtering to remove unwanted noise, to provide spectral shaping, or to perform signal detection or analysis. Digital filter applications include signal preconditioning, band selection, and low pass filtering. These functions are provided by two types of filters i.e. Finite impulse response (FIR) filters and Infinite impulse response (IIR) filters.

The basic characteristics of Finite Impulse Response (FIR) filters are Linear phase, high filter order (more complex circuits) and Stability. The basic characteristics of Infinite Impulse Response (IIR) are Non-linear phase, Low filter order (less complex circuits) and Resulting digital filter has the potential to become unstable. IIR filters have much better frequency response than FIR filters of the same order. Unlike FIR filters, their phase is not linear which can create a problem to the systems which need phase linearity. On the other hand, if the linear phase characteristic is not required, the use of IIR filters is an excellent solution [2].

2. IIR FILTER

IIR filters are digital filters with infinite impulse response. Unlike FIR filters, they have the feedback (a recursive part of a filter) and are known as recursive digital filters. For this reason IIR filters have much better frequency response than FIR filters of the same order. When the linear phase characteristic is not important, the use of IIR filters is an excellent solution [2].

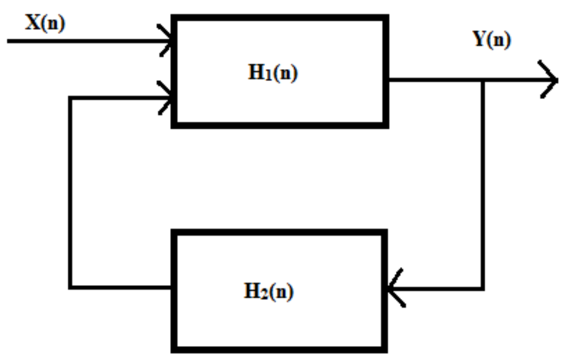

Fig. 2 Block Diagram of IIR Filter

Fig. 2 shows the representation of IIR Filter in which it consist a feedback element which provides a good output response. There are many techniques for designed of digital filter having a infinite duration impulse response. Thus the design of an IIR filter involves design of a digital filter in the analog domain and transforming the design into the digital domain. The system function describing an analog filter may be written as:

$$H_a(s) = \frac{\sum_{k=0}^{M} b_k s^k}{\sum_{k=0}^{M} a_k s^k} \qquad (1)$$

Where a_k and b_k are the coefficient of filter. The system function $H_a(s)$ can also be written by its impulse response as:

$$H_A(s) = \int_{-\infty}^{+\infty} h(t) e^{-st} \qquad (2)$$

For the IIR Filter design it is necessary to find to find suitable values for the coefficients a_k and b_k so that the desired frequency response is obtained. To design digital IIR filter first we have filter specification. The filter specification gives the function of the filter (low pass, high pass, band pass, band reject filter) and the desired performance. After filter specification we have to calculate the coefficient a_k and b_k and realize the structure to analyze finite word length and then hardware and software implementation[2].There are four types of basic prototype filter from which we design IIR filter[4].
The Butterworth filter has a maximally flat response, i.e, No pass bands ripples and roll-off of minus 20db per pole [6]. Low pass Butterworth filters are all-pole filter characterized by the magnitude-squared frequency response

$$|H(\Omega)|^2 = \frac{1}{1+(\Omega/\Omega_C)^{2N}} = \frac{1}{1+\varepsilon^2(\Omega/\Omega_P)^{2N}} \qquad (3)$$

Where N is the order of the filter, Ω_c is its -3dB frequency, Ω_p is the pass band edge frequency, and $1/1+(\varepsilon^2)$ is the band edge value of $|H(\Omega)|^2$.

There are two types of Chebyshev filters. Chebvshev1 filters are all-pole filters that exhibit equiripple behavior in the pass band and a monotonic characteristic in the stop band. The Chebyshev2 filters contain both poles and zeros and exhibits an equiripple behavior in the stop band and a monotonic characteristic in the pass band. The zeros of type 2 filters lie on the imaginary axis in the s plane. The magnitude squared of the frequency response characteristic of a Chebyshev1 filter is given as

$$|H(\Omega)|^2 = \frac{1}{1+\varepsilon^2 T_N^2(\Omega/\Omega_p)} \qquad (4)$$

Where Ɛ is the parameter related to the pass band ripple and $T_N(x)$ is the *Nth* order Chebyshev polynomial defined as

$$T_N(x) = \begin{cases} \cos(N\cos^{-1}x), & |x| \le 1 \\ \cosh(N\cos^{-1}x), & |x| > 1 \end{cases} \qquad (5)$$

The magnitude squared of the frequency response characteristic of a Chebyshev2 filter is given as

$$|H(\Omega)|^2 = \frac{1}{1+\varepsilon^2\left[\frac{T_N^2(\Omega_P/\Omega_S)}{T_N^2(\Omega_S/\Omega)}\right]} \qquad (6)$$

Where $T_N(x)$ is the *Nth* order chebyshev polynomial and Ω_s is the stop band frequency.
Elliptic or Cauer filters exhibit equiripple behavior in both the pass band and the stop band. This class of filters contains both poles and zeros and is characterized by the magnitude –squared frequency response.

$$|H(\Omega)|^2 = \frac{1}{1+\varepsilon^2 U_N\left(\Omega/\Omega_p\right)} \quad (7)$$

Where U_N is the Jacobian Elliptic function of order N and ε is the parameter related to the pass band ripple [3].

3. IIR DESIGN SIMULATION

The Magnitude response and the group delay of the band pass filter using Butterworth, Chebyshev1, Chebyshev2 and Elliptic filters is shown in following Fig. for different filter order i.e. N=8, 16, 32. The cut off frequency used in the frequency range of SONAR (1kHz<Fc<10kHz). So we used Fc_1=1kHz, Fc2=10kHz and the sampling frequency Fs=30kHz.

The combined Magnitude response and the group delay of the band pass filter using Butterworth filter, Chebyshev1 filter, Chebyshev2 filter and Elliptic filter for N=8, Fc_1=1kHz, Fc_2=10kHz and Fs=30kH is shown in Fig.3 and Fig.4

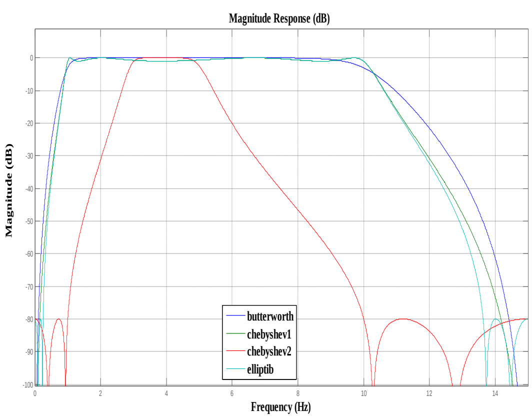

Fig.3 Comparison of magnitude response of different filters for N=8

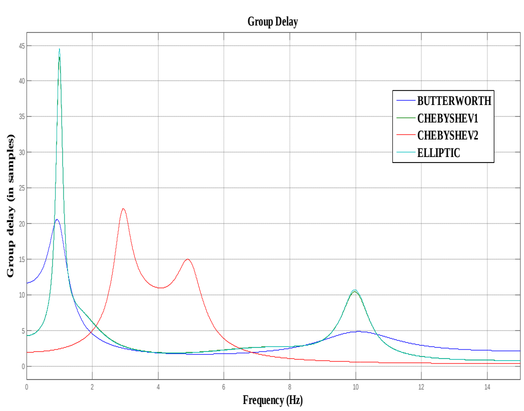

Fig.4 Comparison of Group delay of different filters for N=8

The combined Magnitude response and the group delay of the band pass filter using Butterworth filter,Chebyshev1 filter,Chebyshev2 filter and Elliptic filter for N=16, Fc_1=1kHz,Fc_2=10kHz and Fs=30kH is shown in Fig.5 and Fig.6.

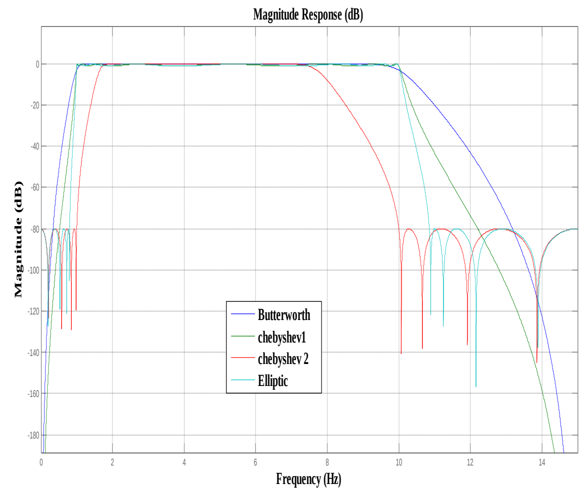

Fig. 5 Comparison of magnitude response of different filters for N=16

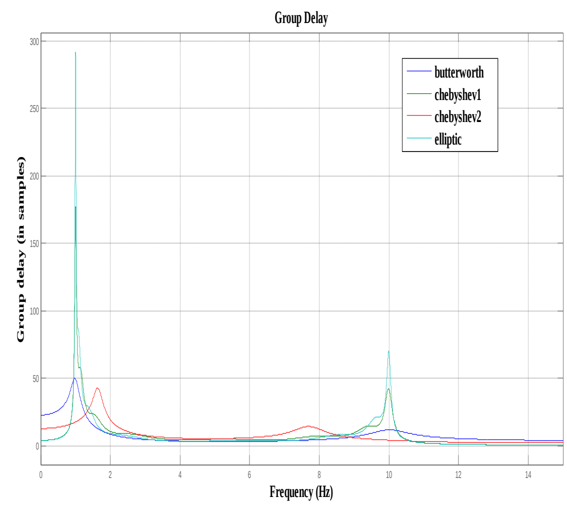

Fig. 6 Comparison of Group delay of different filter for N=16

The combined magnitude response and the group delay of the band pass Butterworth, Chebyshev1, Chebyshev2 and Elliptic filters for N=32, Fc_1=1kHz,Fc_2=10kHz and Fs=30kH is shown in Fig.7 and

Fig.8

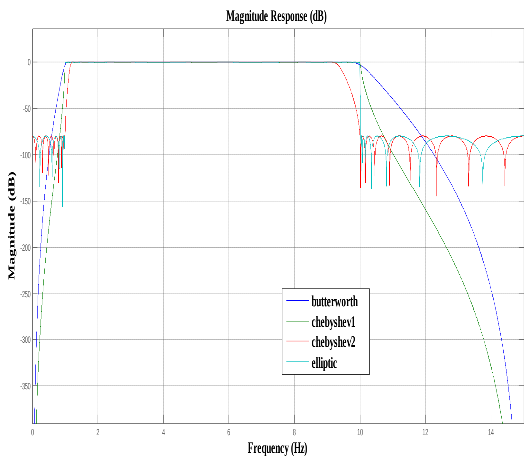

Fig. 7 Comparison of magnitude response of different filters for N=16

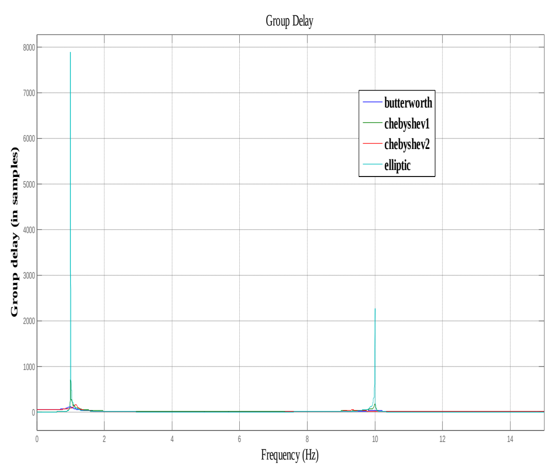

Fig. 8 Comparison of Group delay of different filters for N=32

4. COMPARITIVE ANALYSIS

Comparison of different filter techniques in terms of filter parameters by varying the order can be observed from the different magnitude response of the band pass filter is shown in Table.1.

The comparative analysis of transmission width, Pass band attenuation, Stop band attenuation with changing the order for designing of different prototype filter is shown in terms of bar chart. Fig. 9 shows the transmission width of different filter with different order. Fig. 10 shows the stop band attenuation of different filter with different order. Fig. 11 shows the pass band attenuation of different filter with different order.

Table. 1 Comparison of different filters with following parameters

Order	Prototype-e filter	Transit-ion Width	Stop Band Attenua tion-on	Pass band Attenua tion-on
N=8	Butterworth	0.86243	-72.18069	-0.015434
	Chebys-hev1	0.82397	-84.7213	-0.99314
	Chebys-hev2	1.08859	-80.3322	-0.149095
	Elliptic	0.26879	-78.8772	-0.99299
N=16	Butterworth	0.86426	-146.298	-0.02666
	Chebys-hev1	0.84778	-176.6163	-0.02767
	Chebys-hev2	0.64819	-79.03294	-0.02767
	Elliptic	0.18677	-78.7414	--0.97905
N=32	Butterworth	0.85510	-279.493	-0.61894
	Chebys-hev1	0.85144	-363.808	-0.81852
	Chebys-hev2	0.16845	-80.6327	-0.02266
	Elliptic	0.00732	-80.0565	-0.02266

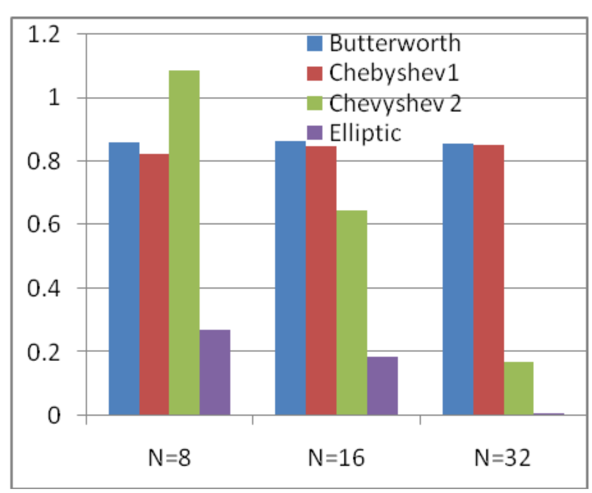

Fig.9 Transmission width (Hz) of different filter with different order

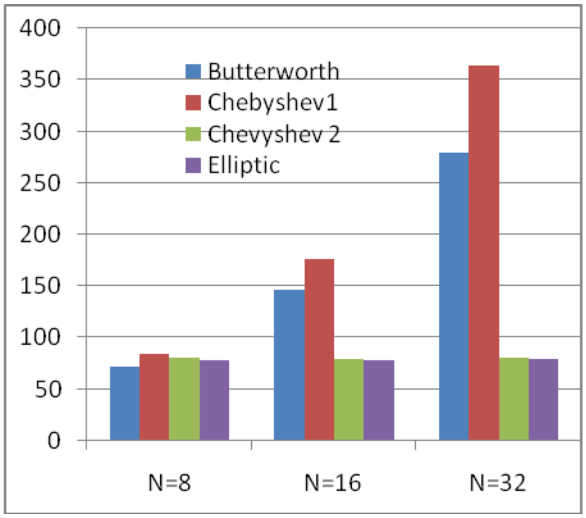

Fig.10 Stop band attenuation (-dB) of different filter with different order

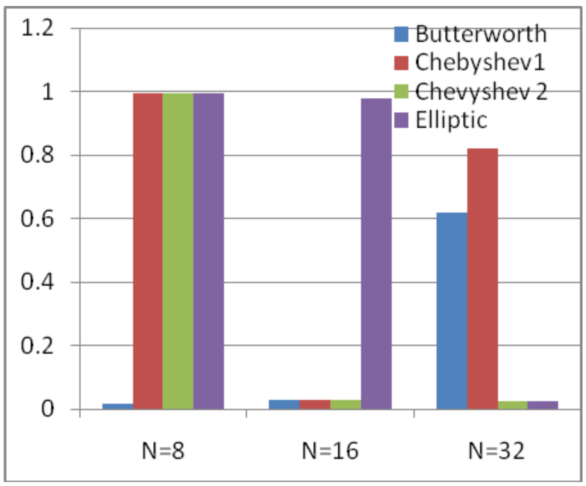

Fig.11 Pass band attenuation (-dB) of different filter with differe order

5. CONCLUSION

From observation and comparison of different parameters of filters we analyzed that the response of Butterworth filter is good at lower order in terms of its magnitude response and group delay. As we increase the order the performance of elliptic filter is getting better continuously as compared to other filters. From observation we can see that the response of chebyshecv1 filter in terms of its stop band attenuation is more as compared to others. As we increase order the transition width of chebyshev2 getting better and stop band attenuation and pass band attenuation is also getting better.

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