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Comparison and Implementation of Different Types of IIR Filters for Speech Signal Analysis

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***Abstract*—In the field of Digital signal processing, the filter plays a vital role to remove unwanted component and to keep only the desired signal. During the transmission of speech signal through the channel, noise which is random in nature will get superimpose with the original signal and to remove those noise from the signal, and also to extract the useful information from the signal, filters are used. Various applications of filter include for example: in audio processing, video processing, image enhancement, pattern recognition, and in the field of biomedical signal processing to analyze heart related problems such ECG signal, to analyze brain related problems such EEG signal and EMG signal filtering is used for extracting useful information from muscles. For military applications radar and sonar processing is used. In the field of instrumentation/control, data compression, noise reduction is possible by using filter. In this paper, comparison of different digital filter such as Butterworth filter, Chebyshev type I & II filter and Elliptic filter have been experimented by using MATLAB software. In this paper, low pass, high pass, band pass and band stop of the above mentioned filter is experimented and also impulse responses, magnitude responses, phase responses and pole-zero plot of Butterworth, Chebyshev type I & II filter and Elliptic filter is observed for Speech Signal Analysis.**

Keywords*— IIR Filter, Butterworth, Chebyshev type I and Elliptic filter, Impulse response, Magnitude response, Phase response, Pole-Zero plot.

I. INTRODUCTION

Filters play an important role in the field of analog and digital signal processing and telecommunication systems. As Analog signal processing uses components such resistors, capacitor, inductor which will not produce same result after some years because of the tolerance of components and also the performance changes with respect to the variation in temperature. If slight changes are required in the performances than one has to redesign the entire circuit by changing the values of component and hence digital signal processing is preferred over analog signal processing for example digital signal can be copied several times in a compact disc where the quality of the signal will remain same. Digital signal processing uses elements such as adder, delay and multiplier. Since there are no component tolerances hence identical performances from unit to unit can be observed [1]. Block diagram of digital signal processing system is shown in *Fig 1.1*. Digital filters can be classified into two categories: IIR filter and FIR filter. IIR filter is basically used to convert analog filter into a digital filter. A realizable IIR digital filters are characterized by the following recursive equation:

$$y(n) = \sum_{k=0}^{N} a_k\, x(n-k) - \sum_{k=1}^{M} b_k\, y(n-k) \qquad (1)$$

IIR filter is called recursive filter because the present values of output depend not only on the present and past values of input but also on the past values of output. IIR filter requires less filter coefficients as compared to FIR filter for the same specifications and hence IIR filters is used if sharp cutoff frequency and high throughput is required [2]. IIR filter is used where linear phase characteristics is not required, and IIR filter is used in low-power communication system.

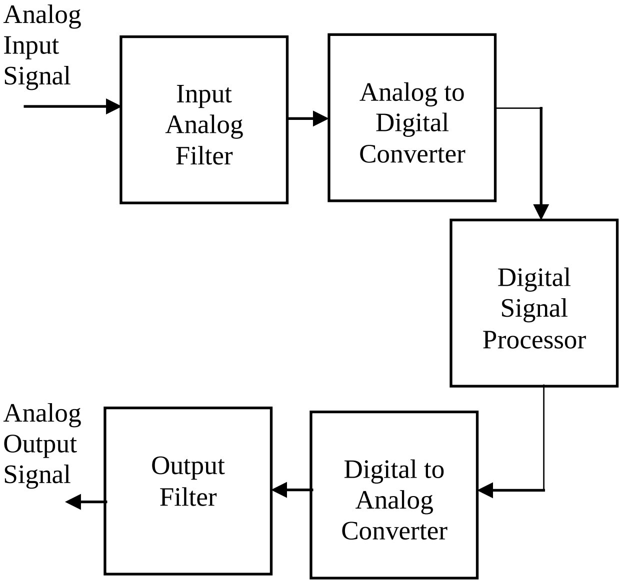

Fig 1.1 Block diagram of a digital signal processing system

The IIR filter has some advantages over FIR filter which is given below:

1. For the same filter specifications, IIR filter requires lower order as compared to FIR filter.
2. In the stopband of IIR filter it contains less number of side lobes.
3. IIR filters consist of zeros and poles, and it requires less memory as compared to FIR filter.
4. The computational efficiency of IIR filter is high with short delays.

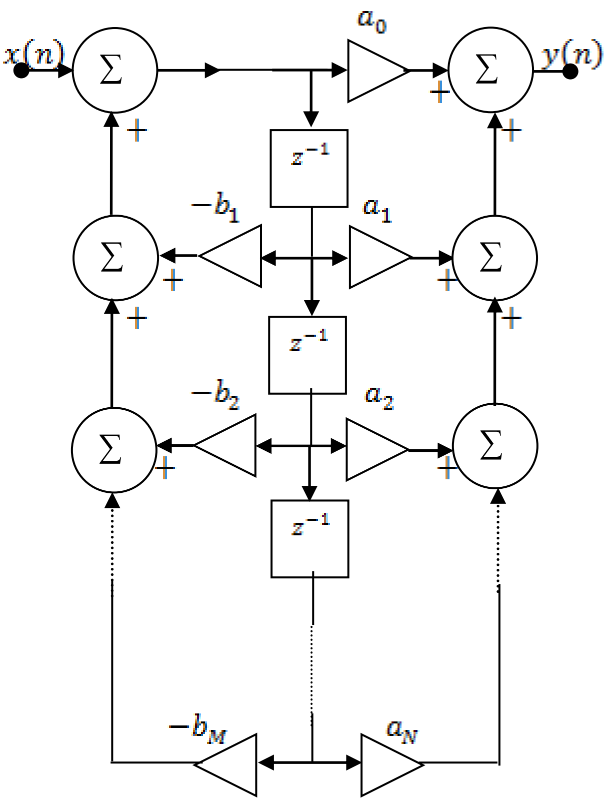

Fig 1.2 Realization of IIR Filter

II. BUTTERWORTH FILTER

The magnitude squared function of Butterworth low pass filter is given in the following equation.

$$|H(j\Omega)|^2 = \frac{1}{1+\left(\frac{\Omega}{\Omega_p}\right)^{2N}} \quad (2)$$

$$N = \frac{1}{2}\frac{\log_{10}\left(\frac{\frac{1}{A_2{}^2}-1}{\frac{1}{A_1{}^2}-1}\right)}{\log_{10}\left(\frac{\Omega_s}{\Omega_p}\right)} \quad (3)$$

where N is order of the filter and Ω_p is the 3dB cutoff frequency and to determine the order of Butterworth filter equation (3) is used. The magnitude response of Butterworth low pass filter decreases monotonically as the frequency increases. Butterworth filter has no ripples in the passband and stopband. The width of the transition band is more in Butterworth filter compared to Chebyshev filter [3] & [4].

III. CHEBYSHEV FILTER

There are two types of Chebyshev filter called as Chebyshev Type-I filter and Chebyshev Type-II filter. The magnitude response of Chebyshev Type-I filter has equal ripples in the passband and monotonically decreasing response in the stopband whereas the Chebyshev Type-II filter has equal ripples in the stopband and monotonically decreasing response in the passband. The magnitude squared function of Chebyshev Type-I low pass filter is given in the following equation.

$$|H(j\Omega)|^2 = \frac{k}{1+\mu^2 C_N^2\left(\frac{\Omega}{\Omega_p}\right)} \quad (4)$$

$$C_N\left(\frac{\Omega}{\Omega_p}\right) = \begin{cases} \cos\left[N\cos^{-1}\left(\frac{\Omega}{\Omega_p}\right)\right], & |\Omega| \le \Omega_p \\ \cosh\left[N\cosh^{-1}\left(\frac{\Omega}{\Omega_p}\right)\right], & |\Omega| \ge \Omega_p \end{cases} \quad (5)$$

$$C_N\left(\frac{\Omega_s}{\Omega_p}\right) = \frac{1}{\mu}\sqrt{\left(\frac{1}{(A_2)^2}-1\right)} \quad (6)$$

$$N = \frac{\cosh^{-1}\left(C_N\left(\frac{\Omega_s}{\Omega_p}\right)\right)}{\cosh^{-1}\left(\frac{\Omega_s}{\Omega_p}\right)} \quad (7)$$

$$N = \frac{\cosh^{-1}\left(\sqrt{\left(\frac{\frac{1}{(A_2)^2}-1}{\frac{1}{(A_1)^2}-1}\right)}\right)}{\cosh^{-1}\left(\frac{\Omega_s}{\Omega_p}\right)} \quad (8)$$

where $C_N(\Omega/\Omega_p)$ is a Chebyshev polynomial which exhibits equal ripple in the passband, N is the order of the polynomial and N also represents order of the filter and μ is represents passband ripple. Order of Chebyshev Type-I filter and Chebyshev Type-II filter can be found by using equation (7) and (8). The Order of Chebyshev filter is less as compared to Butterworth filter for the same specifications and hence we require less components to implement Chebyshev filter.

IV. ELLIPTIC FILTER

Elliptic filter has equal ripple in the passband as well as in the stopband. Order of Elliptic filter is less as compared to Butterworth and Chebyshev filter, also it has small transition band. Elliptic filter is difficult to design as it contains both poles and zeros. Elliptic filter is used for removing noise in ECG signal. The magnitude squared response of the Elliptic filter is given in the following equation.

$$|H(j\Omega)|^2 = \frac{k}{1+\mu^2 C_N^2(\Omega)} \quad (7)$$

where $C_N(\Omega)$ is a Chebyshev rational function and N is the order of the filter. Elliptic filter also called Cauer filter.

V. SIMULATION RESULTS

To design and implement digital filter such low pass, high pass, band pass and band stop with different filter

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specifications such as Butterworth, Chebyshev Type-I, Chebyshev Type-II and Elliptic filter. we need to choose suitable frequency range for designing low pass, high pass, band pass and band stop filters. Table I indicates the frequency specification for designing various types of IIR filter.

TABLE I INDICATES FREQUENCY SPECIFICATIONS OF FILTER DESIGN

FILTER TYPE	Kp in dB	Ks in dB	PASSBAND FREQUENCY IN Hz		STOPBAND FREQUENCY IN Hz	
LPF	3	50	1200		2500	
HPF	3	50	2500		1200	
BPF	3	50	1000	1800	600	2500
BSF	3	50	600	2500	1000	1800

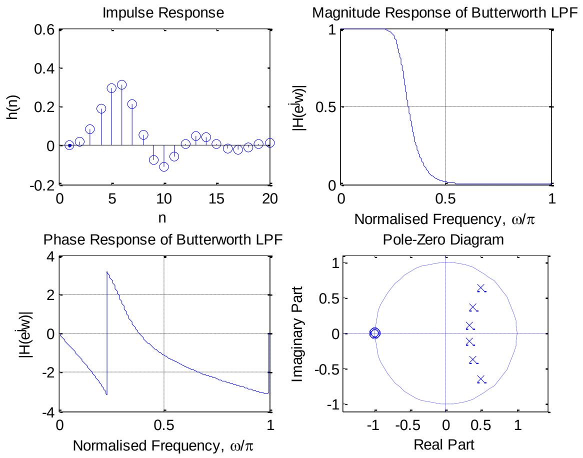

Fig 1.3 Butterworth Low Pass Filter

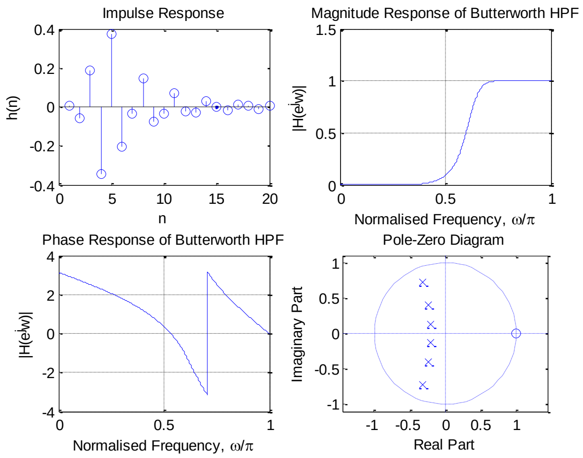

Fig 1.4 Butterworth High Pass Filter

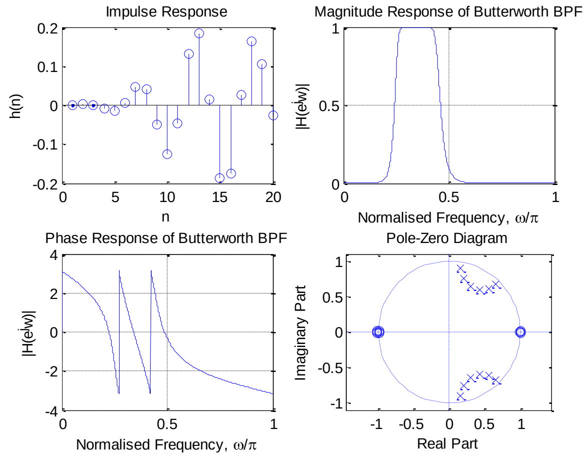

Fig 1.5 Butterworth Band Pass Filter

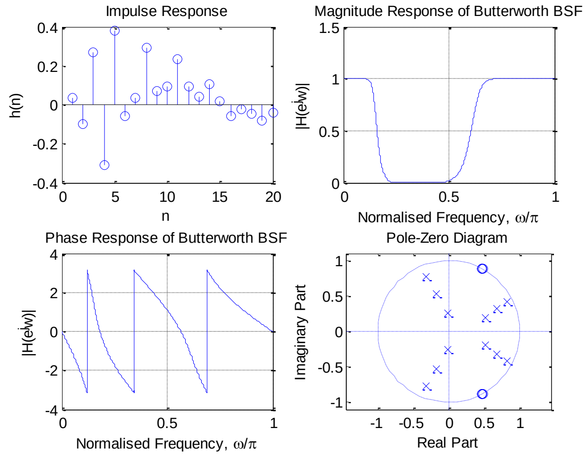

Fig 1.6 Butterworth Band Stop Filter

The impulse response, magnitude response, phase response and pole-zero plot of Butterworth LPF, HPF, BPF and BSF is shown in Fig 1.3, Fig 1.4, Fig 1.5 and Fig 1.6 respectively.

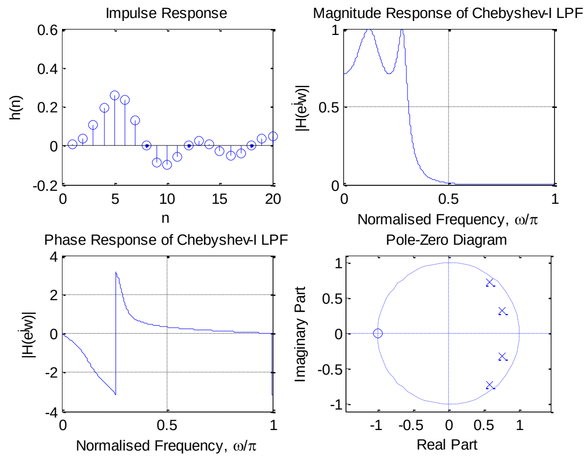

Fig 1.7 Chebyshev Type-I Low Pass Filter

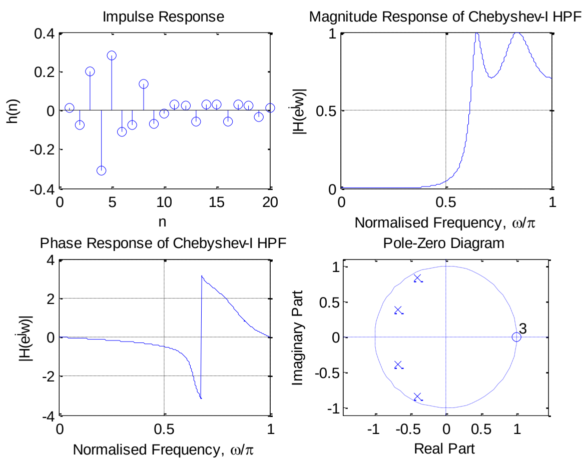

Fig 1.8 Chebyshev Type-I High Pass Filter

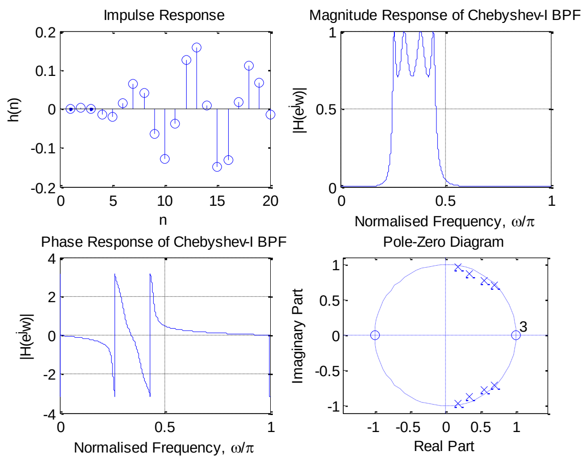

Fig 1.9 Chebyshev Type-I Band Pass Filter

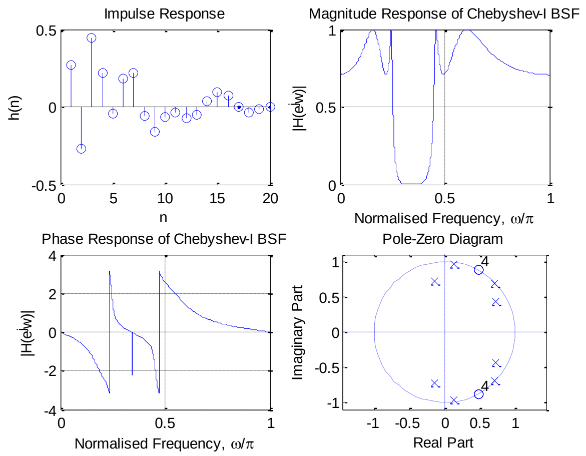

Fig 1.10 Chebyshev Type-I Band Stop Filter

The impulse response, magnitude response, phase response and pole-zero plot of Chebyshev Type-I LPF, HPF, BPF and BSF is shown in Fig 1.7, Fig 1.8, Fig 1.9 and Fig 1.10 respectively.

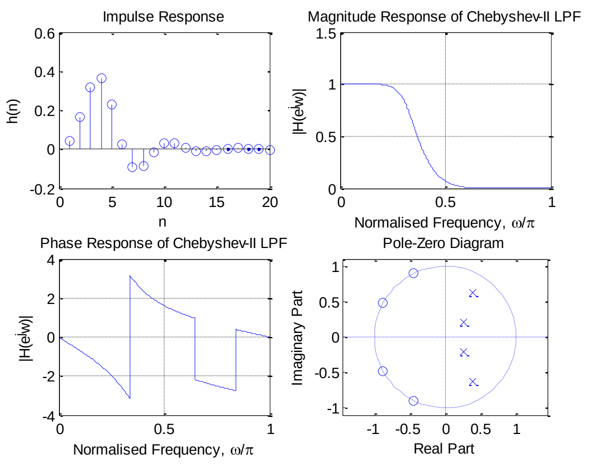

Fig 1.11 Chebyshev Type-II Low Pass Filter

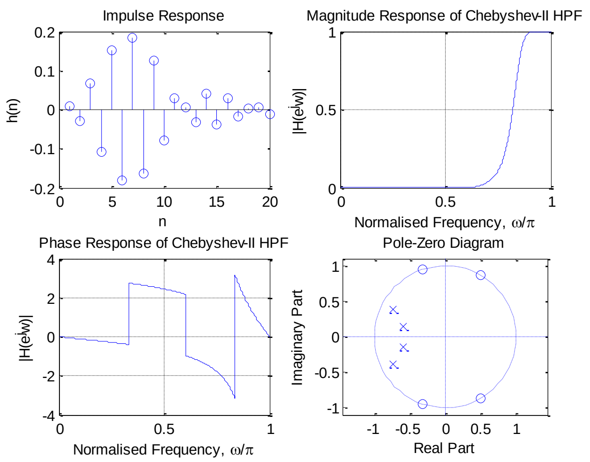

Fig 1.12 Chebyshev Type-II High Pass Filter

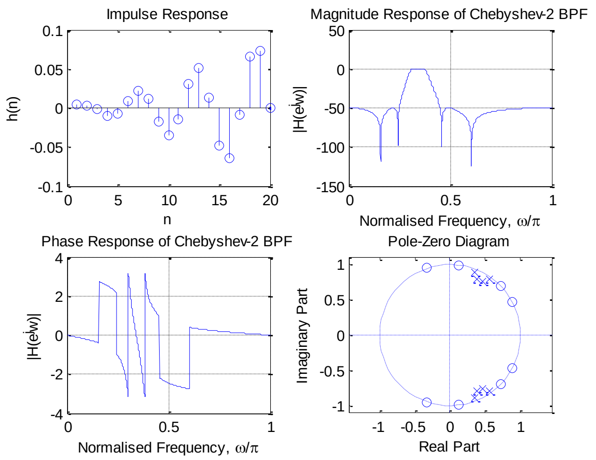

Fig 1.13 Chebyshev Type-II Band Pass Filter

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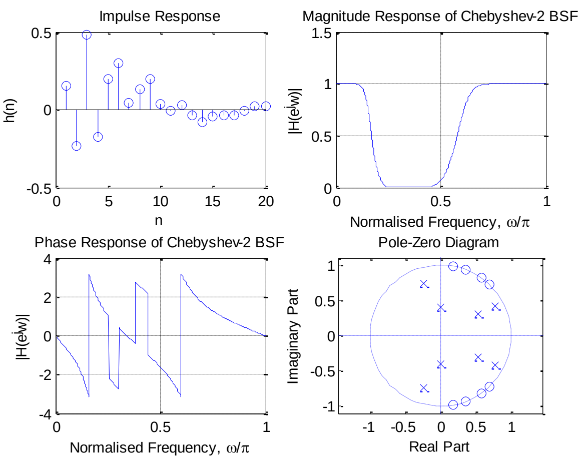

Fig 1.14 Chebyshev Type-II Band Stop Filter

The impulse response, magnitude response, phase response and pole-zero plot of Chebyshev Type-II LPF, HPF, BPF and BSF is shown in Fig 1.11, Fig 1.12, Fig 1.13 and Fig 1.14 respectively.

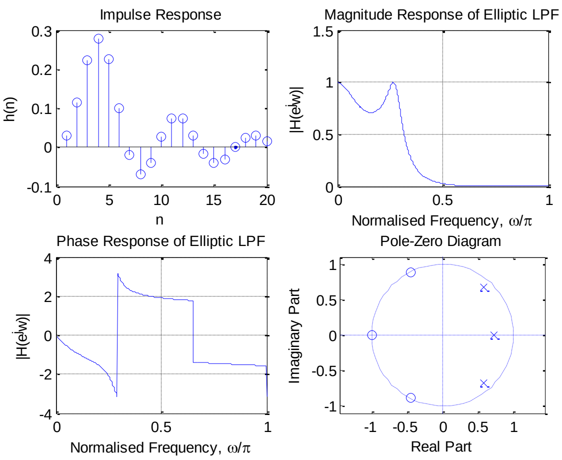

Fig 1.15 Elliptic Low Pass Filter

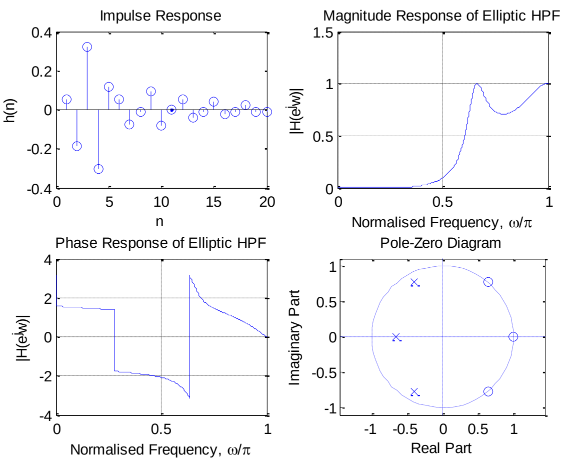

Fig 1.16 Elliptic High Pass Filter

The impulse response, magnitude response, phase response and pole-zero plot of Elliptic LPF, HPF, BPF and BSF is shown in Fig 1.15, Fig 1.16, Fig 1.17 and Fig 1.18 respectively.

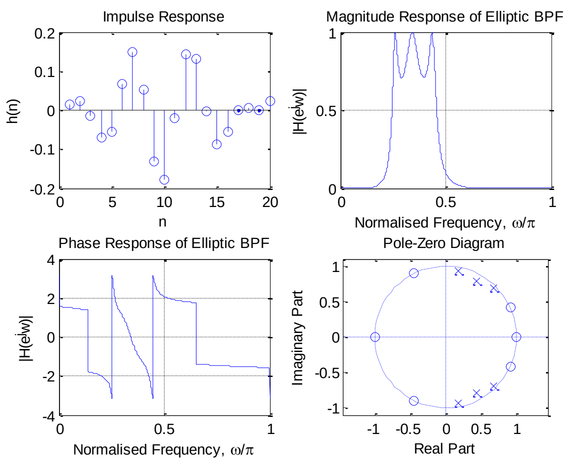

Fig 1.17 Elliptic Band Pass Filter

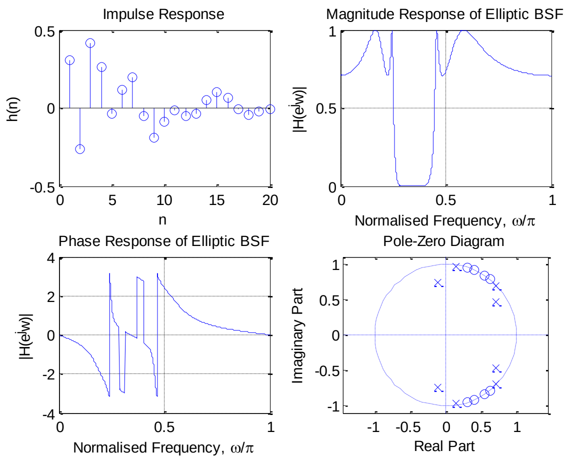

Fig 1.18 Elliptic Band Stop Filter

VI. CONCLUSION

From the magnitude response of LPF, HPF, BPF and BSF of Butterworth filter, we can see that there is no attenuation in the pass-band and stop-band as shown in Fig 1.3, Fig 1.4, Fig 1.5 and Fig 1.6. Butterworth filter has smooth decreasing response for increasing the frequency and it has good phase response but the order required for designing this filter is 6 and it can also be noted from the above mentioned figures, that the Butterworth filter consists no ripples. Fig 1.7, Fig 1.8, Fig 1.9, Fig 1.10, Fig 1.11, Fig 1.12, Fig 1.13 and Fig 1.14 shows magnitude response of LPF, HPF, BPF and BSF of Chebyshev Type-I & II filter contains ripples and transition gap between passband and stopband region is small as compared to Butterworth filter and also order required for designing this filter is 4 which is less than the order required for Butterworth filter and hence its advantages to use Chebyshev Type-I & II filter because it requires less components and less computational cost. Fig 1.15, Fig 1.16, Fig 1.17 and Fig 1.18 shows magnitude response of LPF, HPF, BPF and BSF of Elliptic filter has order 3 for the same specifications which is less than the above mentioned filter, transition region is also much less than the Butterworth and Chebyshev Type-I & II filter. Elliptic filter contains ripples in both passband and stopband [5], and hence it is practically difficult to design.

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In this paper, among all above the specifications the Chebyshev Type-I & II filter are the best filter in terms of order, computational complexity and in terms of economic purpose. The impulse response, magnitude response, phase response and pole-zero plot is experimented & implemented by using MATLAB.

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