Design IIR Filter using MATLAB

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***Abstract* – in Digital Signal Processing (DSP), most widely used filters are two types IIR(infinite impulse response) and FIR(finite impulse response),in this paper IIR filters designed with Bilinear method. Bilinear Transformation method overcomes the aliasing effect and it is best methods for designing IIR digital filters from reference analog filters due to implicitly and similarity of the frequency response of IIR digital filter to that reference analog filters. This method produces true frequency-to-frequency transformation. IIR filters are designed and analyzed by FDATOOL and the implementation cost has been designed on the basis of filter order, adder, impulse samples, and multiplier.**

***Index terms*– IIR Filter, Bilinear Transformation method, MATLAB, FDATOOL.**

I. INTRODUCTION

Filtering is a process by which the frequency spectrum of a signal can be modified, reshaped or manipulated. Within few days Digital Signal Processing(DSP) has grown to important both technologically and theoretically.Two important types of system present in TheDSP. First type of system is signal representation frequency domain and hence it is known as spectrum analyzer. The second type system performs signal filtering in time domain is known as Digital filter, Digital filtering is one of the powerful tools of DSP. Digital filters have capability to performance specifications that would, at best, be extremely difficult, if it isnot impossible, to achieve with an Analog implementation.

Two types of filters are used in DSP system Analog and Digital filters are classified either as Finite duration impulse response (FIR) filters or Infinite duration impulse response (IIR) filters. In FIR the impulse response sequence is of finite duration, i.e. it has finite number of nonzero terms. In IIR filters can often provide a much better performance and less implementation cost than FIR filters. There Lowpass IIR (Infinite impulse response) filters are designed with Bilinear Transformation method.this method very easy and simple design with filters and eliminating Aliasing effect that is present in Impulse invariance method, it is big drawback of Impulse invariance method. In impulse response method, the derived IIR Digital filter has exactly the same impulse response as the original analog filter for continuous time $t = nT_s$, where T_s is the periodic time.

But in Bilinear method Digital filter has approximately same time domain response as the original analog filter for any value of input.Lowpass IIR filter design with Bilinear transform method is a method of compressing the infinite, straight analogue frequency to a finite one long enough to wrap around the unit circle once only. This is sometimes called frequency warping.In this paper author is designed Lowpass IIR filter with FDAtool in MATLAB, it gives to us that which filter has good efficiency.

II.BILINEAR TRANSFORMATION METHOD

The Bilinear Transformation method overcomes the effect of aliasing.

$$H_A(s) = \frac{b}{s+a} \qquad (1)$$

Equation (1) shows the Analog filter, by Bilinear Transformation methods we can derived Digital filter from Analog filter as

$$H_D(z) = H_A(s)|_{s=(2/Ts)(z-1/z+1)} \qquad (2)$$

Where $H_D(z)$ = Transfer function of Digital filter

$H_A(s)$ = Transfer function of Analog filter

T_s = Sampling period

Here mapping properties of Bilinear transformation will be studied. The relation between the frequency response of derived digital filter and that of original analog filter can be established by examining these mapping properties.

We know that for Bilinear transformation method.

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$$R = \sqrt{\frac{\left(\frac{2}{Ts}+\sigma\right)^2 + w^2}{\left(\frac{2}{Ts}-\sigma\right)^2 + w^2}}$$

$$\Theta = \tan^{-1}\frac{\omega}{(\frac{2}{Ts}+\sigma)} + \tan^{-1}\frac{\omega}{(\frac{2}{Ts}-\sigma)}$$

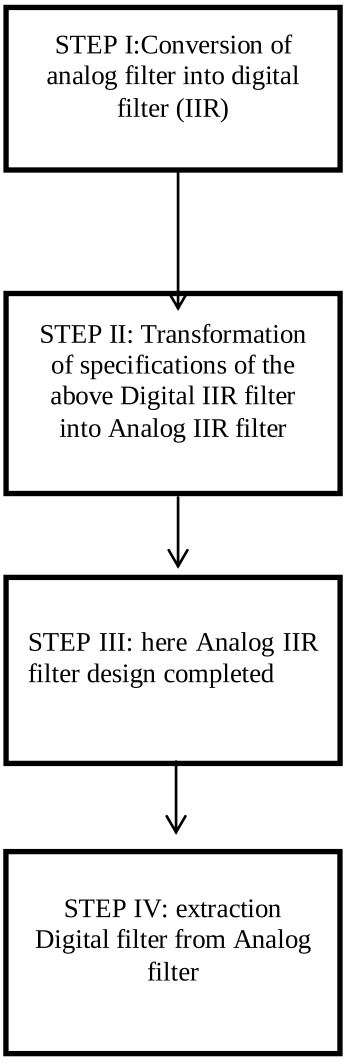
Case I: if $\sigma > o$, then $r>1$, i.e., the bilinear transformation maps the open right-half s-plan onto the region exterior to the unit circle $|z| = 1$ of the z-plane.

Case II: if $\sigma < o$, then $r<1$, i.e., the bilinear transformation maps the open left-half s-plan onto the interior of the unit circle $|z| = 1$ of the z-plane.

Case III: if $\sigma = o$, then $r = 1$, i.e., the bilinear transformation maps the imaginary axis of the s-plane onto the unit circle $|z| = 1$ of the z-plane.

II. FILTER DESIGN

Filter design by the flow chart, this approach of designing the digital filter from analog filter is easy.

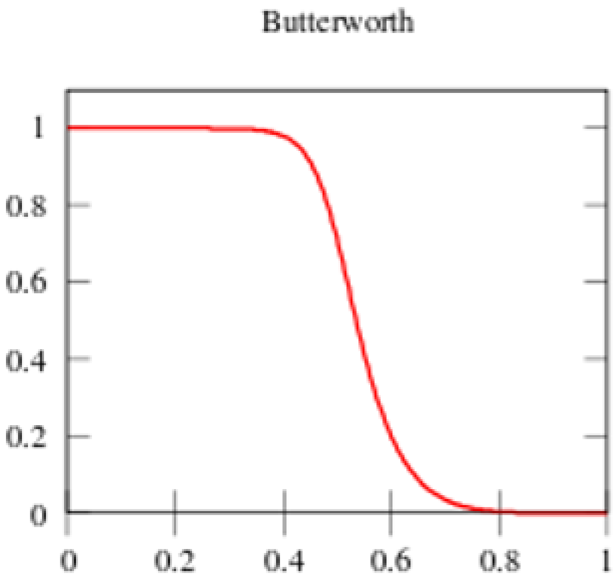


Four types of IIR filter has been designed with the help of FDATOOL in MATLAB. These filters are Butterworth, Chebyshev I, Chebyshev II, Elliptic, these are designed below.

(1) Butterworth filter : Butterworth method for analog filter design plays a very important role because of its simplicity and also because the magnitude characteristics are very nearly ideal near the cut off frequency of high order filter.Butterworth filter is causal in various order, the lowest order being the best in the time domain.

Butterworth or monotonically flat filter has monotonic amplitude frequency response which is maximally flat at zero frequency response and amplitude frequency response decrease logarithmically with increasing frequency.

$$|H(\omega)|^2 = \frac{1}{1+(\frac{\omega}{\omega o})^{\wedge}2n}$$



Butterworth characteristics fig.1

(2) Chebyshev I:chebyshev I filters are all pols filters which are equeripple in the passband and monotonic in stopband.

$$|H(\Omega)| = (1+\epsilon^2 T_N^2(\frac{\Omega}{\Omega p}))^{-1}$$

Where ϵ is a parameter related to the ripple present in the passband.

$$T_N = \cos(N\cos^{-1} x) \qquad |x| \le 1$$

$$\mathrm{Cos}(N\cosh^{-1} x) \qquad |x| \ge 1$$

(3) Chebyshev II: chebyshev II filters contain both zeros and poles. There is equeripple in the stopband and a monotonic behavior in the passband.

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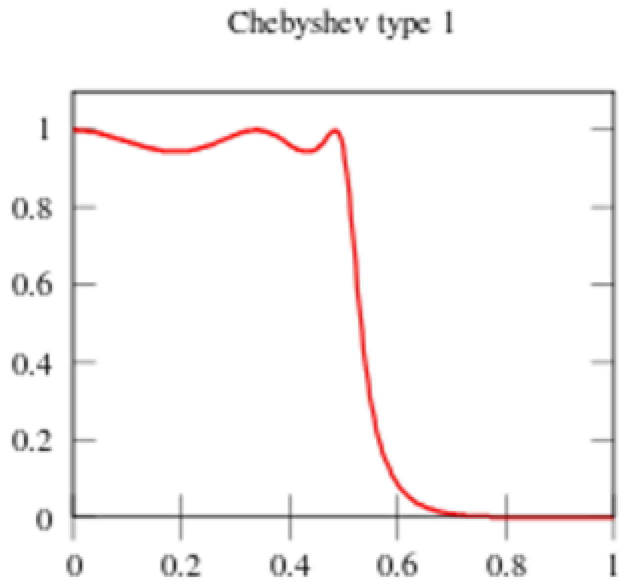
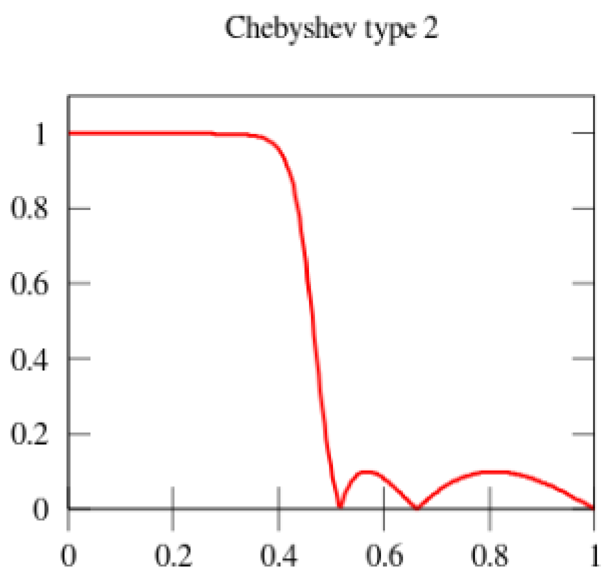


ChebyshevI characteristics fig.2



Chebyshev II characteristics fig.3

(4) Elliptic: Elliptic filter is characterized by equeripple in both passband and stopband. They provide a realization with lowest order for a particular set of condition.

$$|H(j\Omega)| = 10^{-Rp/20} \text{ at } \Omega = 1$$

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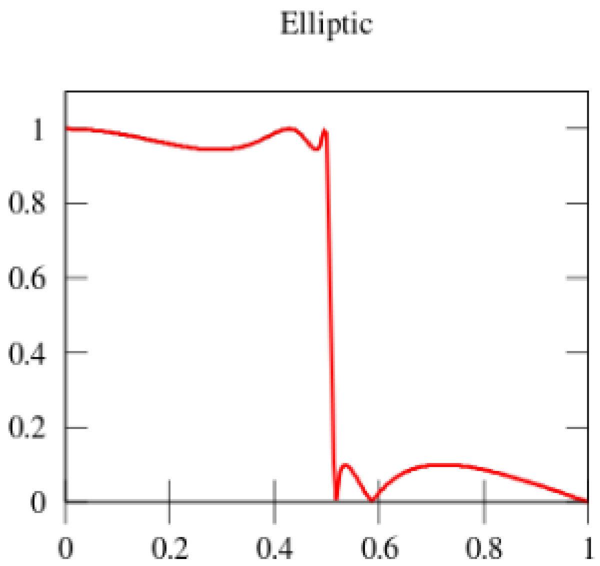

s Elliptic characteristics fig.4

III. SIMULATION RESULT

This program required for designing the IIR Lowpass filters are simulated in MATLAB 8.3. in this paper, the simulation result are shown for the IIR Butterworth, ChebyshevI, ChebyshevII, Elliptic Lowpass filter using the modified Analog to Digital mapping technique coefficients are essential for designing the filter. So, here the coefficients are calculated along the simulations results. The pole-zero diagram shows that the desgned filter is stable. The figure for the phase response, impulse response, magnitude response and pole-zero plot are shown below figure.

Coefficients are shown in table:-

Table I

Filter Name	**Order of Filter**	**Numerator coefficient**	**Denominator coefficient**
Butterworth filter	11	1 2 1	1 -0.216 0.752
	15	1 2 1	1 -1.4127 0.87167
	21	1 2 1	1 -0.4166 0.864

Table II

Types of Filter	**Filters order**	**Number of multiplier**	**Number of Adder**	**Number of state**
Butterworth	21	42	42	21
ChebyshevI	9	18	18	9
ChebyshevII	9	18	18	9
Elliptic	5	10	10	5

In order to illustrate the efficiency of the designed Hardware architecture for the IIR filter.shown in TableII, in Elliptical filter ,filter order, number of multiplier, number of adder, number of states up to 98.56%, 98.56%, 98.56%, 98.56% as compared to butterworth. Whereas in comparison of elliptic filter with chebyshev I and chebyshev II filters order, multiplier, adder, number of states up to 88.24%, 88.24%, 88.24%, 88.24%.

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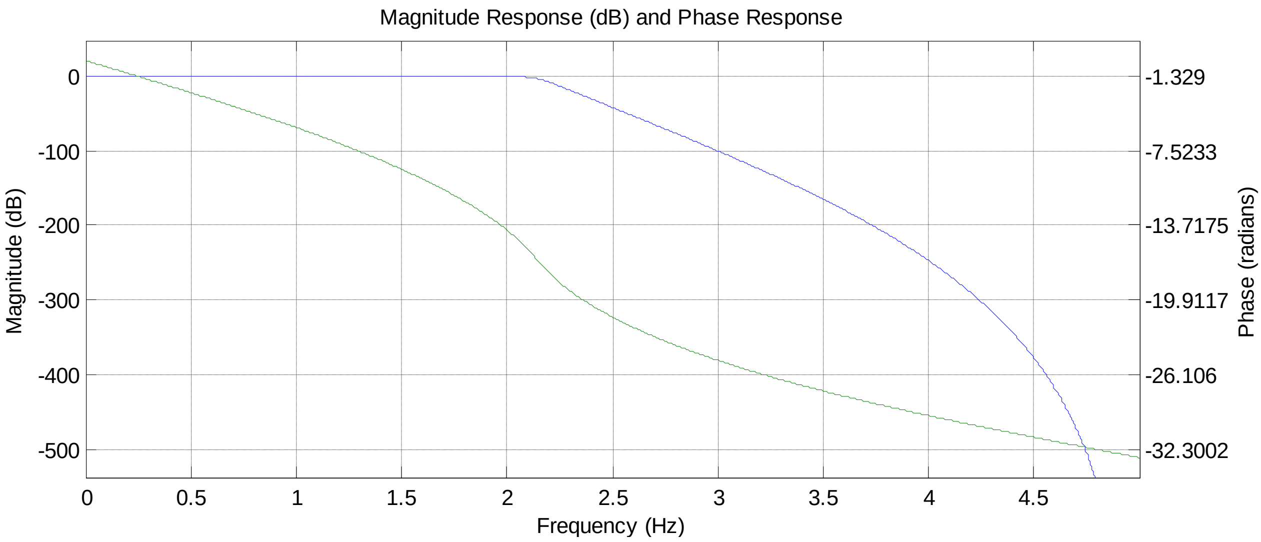
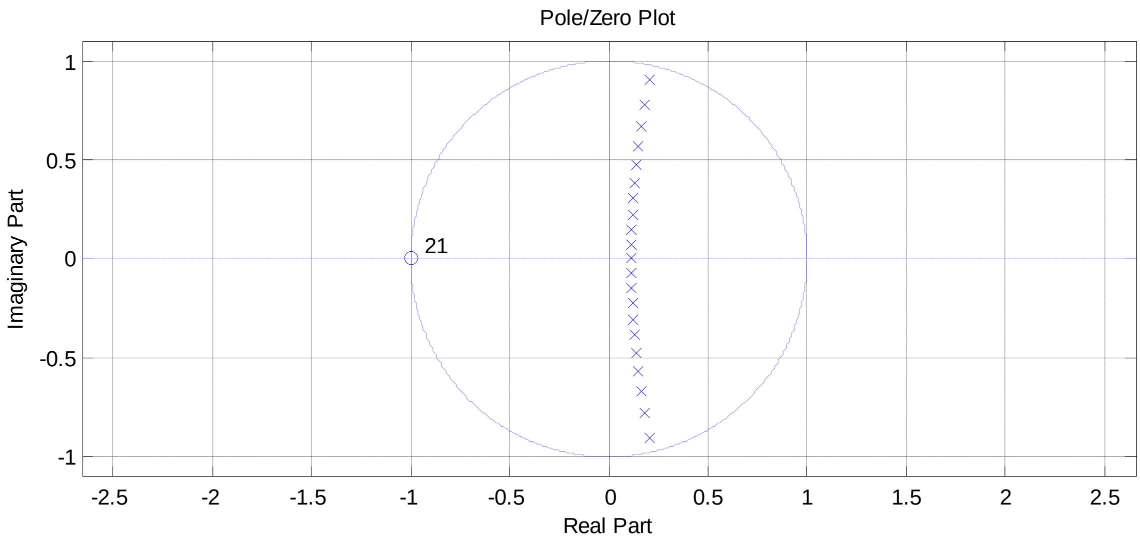
- Butterworth Filter with **21order**





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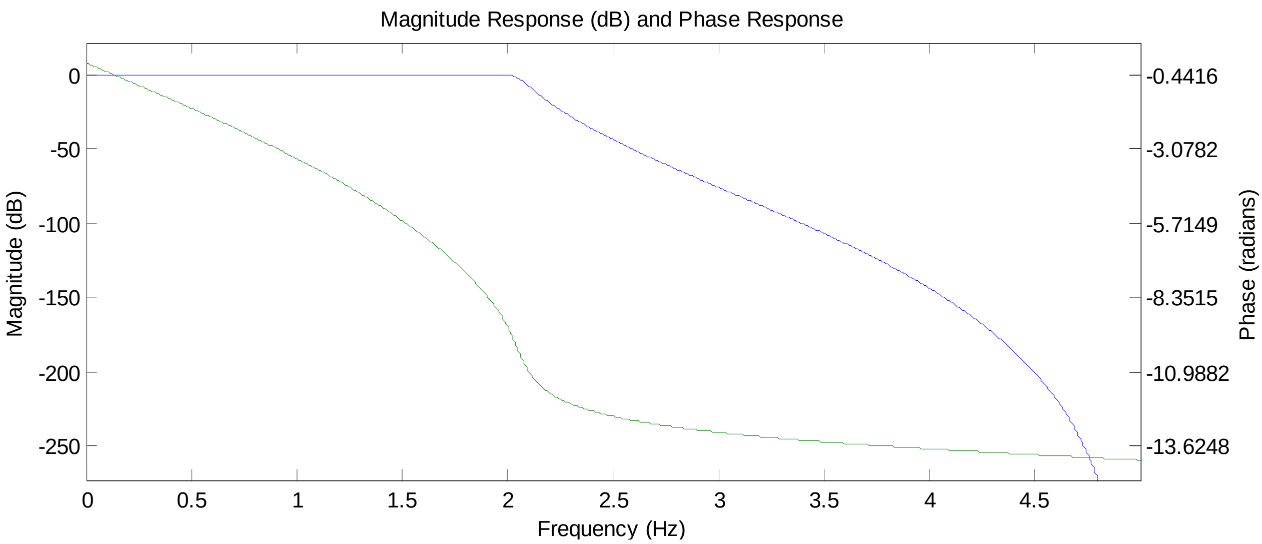
- ChebyshevI Filter **9order**



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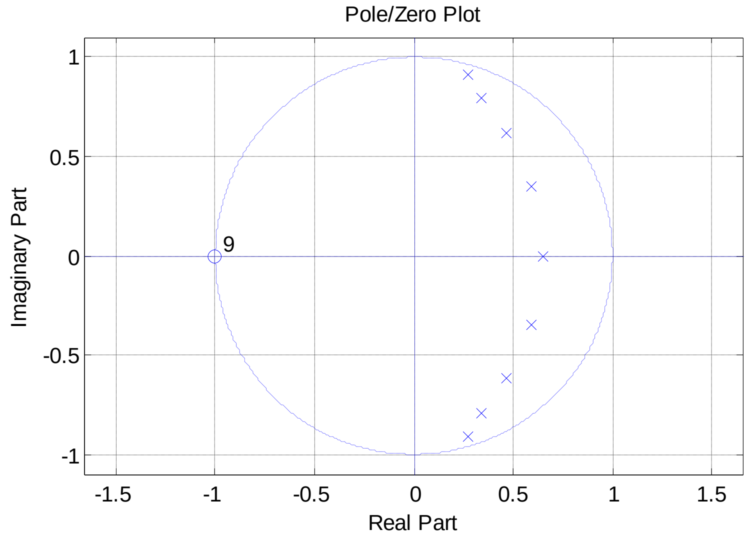
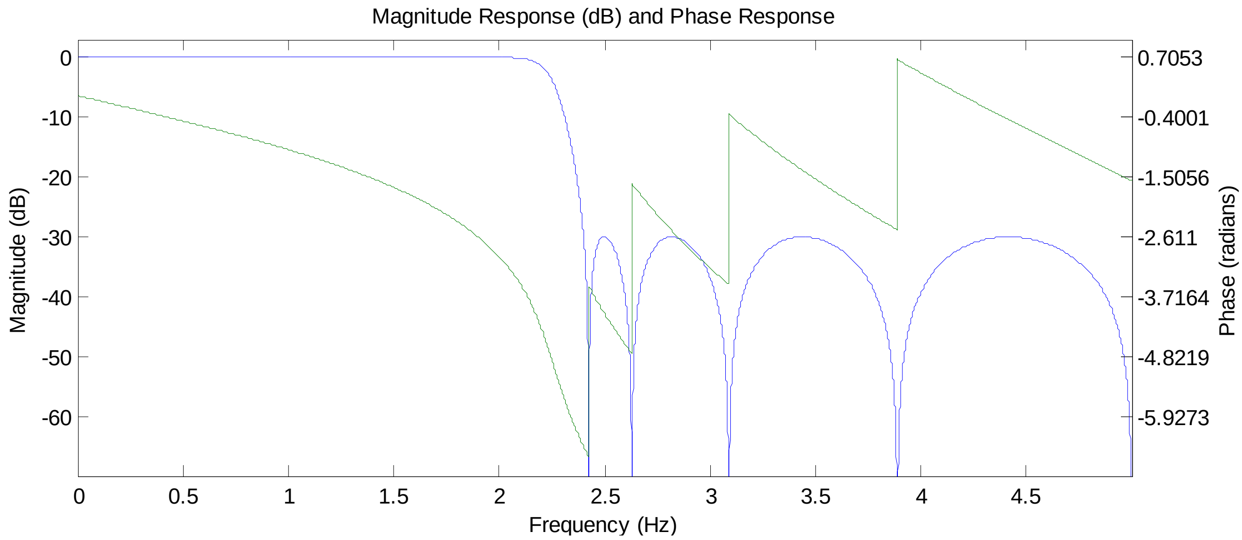
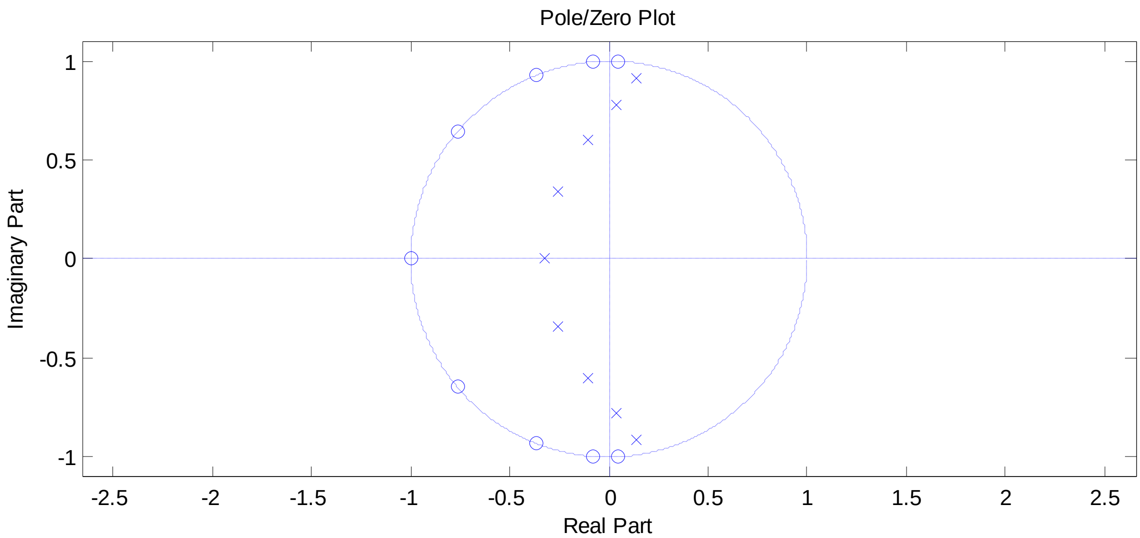


- ChebyshevII Filter **9order**





- Elliptic Filter **5order**

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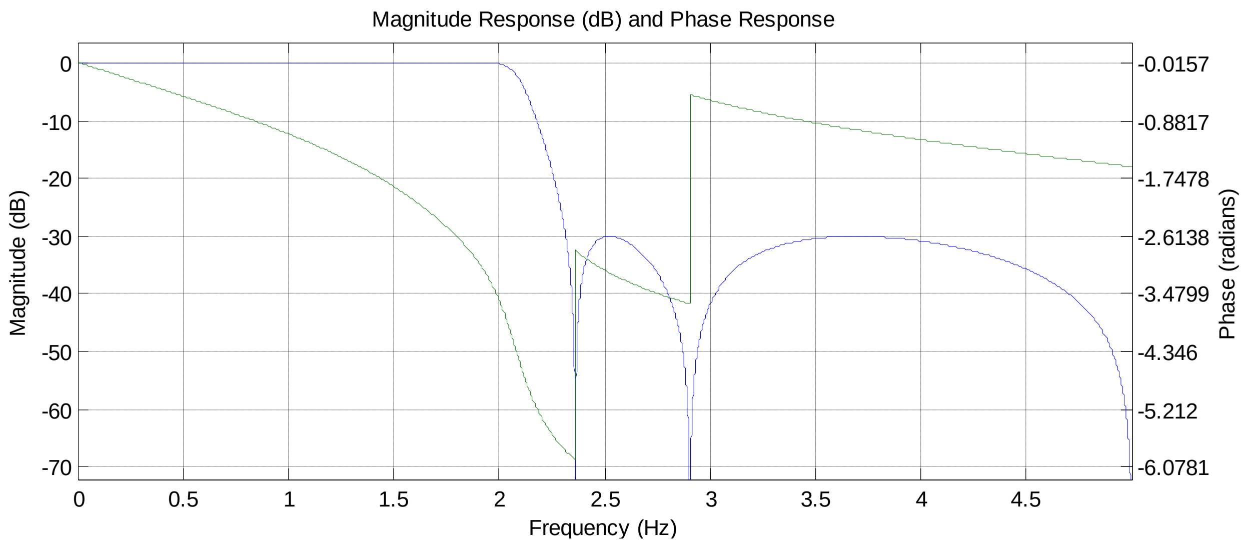

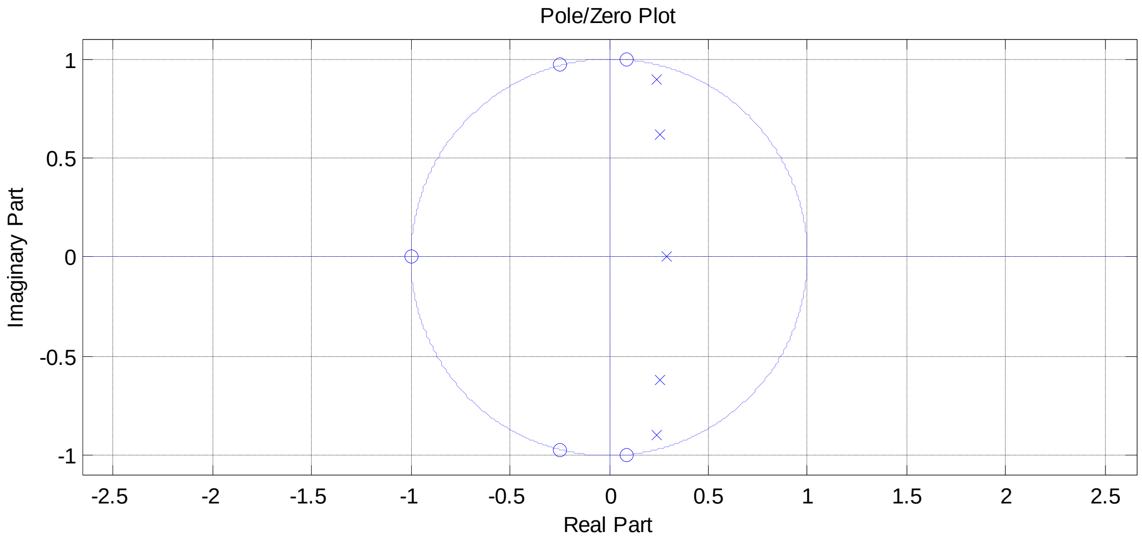

IV. CONCLUSION

The above simulation shows that the designed filter is stable. We can see in pole-zero diagram. Table II shows the efficiency of filter, that shows in same sampling frequency, passband edge frequency, and stopband edge frequency Elliptic filter is more efficient than the butterworth filter and chbyshev filter. Elliptic IIR Filter can offer some important advantages over their substantially lower computational or Hardware complexity.

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