

*EE 338 : Digital Signal Processing*

**Filter Design Report**

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# Assigned Filter Number: 18

## 1 Filter-1 (Bandpass) Details

### 1.1 Un-normalised discrete time filter specifications

Since Filter number is  $< 75$ ,  $m = 18$  and passband will be monotonic

$$q(m) = \lfloor 0.1m \rfloor = 1$$

$$r(m) = m - 10 \cdot q(m) = 18 - 10 = 8$$

$$B_L(m) = 150 + 17 \cdot q(m) + 13 \cdot r(m) = 271$$

$$B_H(m) = B_L(m) + 45 = 316$$

The first filter is given to be a **Band-Pass** filter with passband from  $B_L(m)$  kHz to  $B_H(m)$  kHz. Therefore the specifications are : –

- **Passband** : 271 kHz to 316 kHz
- **Transition Band** : 20 kHz on either side of passband
- **Stopband** : 0-251 kHz and 336-600 kHz ( $\because$  Sampling rate is 1.2 MHz)
- **Tolerance** : 0.15 in magnitude for both Passband and Stopband
- **Passband Nature** : Monotonic
- **Stopband Nature** : Monotonic

### 1.2 Normalized Digital Filter Specifications

Sampling Rate = 1.2 MHz

In the normalized frequency axis, sampling rate corresponds to  $2\pi$

Thus, any frequency( $\Omega$ ) upto 600 kHz ( $= \frac{\text{SamplingRate}}{2}$ ) can be represented on the normalized axis( $\omega$ ) as : –

$$\omega = \frac{\Omega * 2\pi}{\text{SamplingRate}}$$

Therefore, the corresponding normalized digital filter specifications are : –

- **Passband** :  $0.45167\pi$  to  $0.5267\pi$
- **Transition Band** :  $0.033\pi$  on either side of passband
- **Stopband** :  $0-0.41833\pi$  and  $0.56\pi-\pi$
- **Tolerance** : 0.15 in magnitude for both Passband and Stopband
- **Passband Nature** : Monotonic
- **Stopband Nature** : Monotonic

### 1.3 Analog filter specifications using Bilinear Transformation

The bilinear transformation is given as : –

$$\Omega = \tan\left(\frac{\omega}{2}\right)$$

Applying the Bilinear transform to the frequencies at the band-edges, we get : –

$\omega$	$\Omega$
0	0
$0.41833\pi$	0.771
$0.45167\pi$	0.859
$0.5267\pi$	1.088
$0.56\pi$	1.209
$\pi$	$\infty$

Therefore the corresponding analog filter specifications for the same type of analog filter using the bilinear transformation are : –

- **Passband** :  $0.859(\Omega_{P_1})$  to  $1.088(\Omega_{P_2})$
- **Transition Band** :  $0.771$  to  $0.859$  and  $1.088$  to  $1.209$
- **Stopband** :  $0$  to  $0.771(\Omega_{S_1})$  and  $1.208(\Omega_{S_2})$  to  $\infty$
- **Tolerance** :  $0.15$  in magnitude for both Passband and Stopband
- **Passband Nature** : Monotonic
- **Stopband Nature** : Monotonic

### 1.4 Frequency Transformation and Relevant Parameters

We need to transform a Band-Pass analog filter to a Lowpass analog filter. We require two parameters in such a case. We can make use of the Bandpass transformation which is given as : –

$$\Omega_L = \frac{\Omega^2 - \Omega_0^2}{B\Omega}$$

The two parameters in the above equation are  $B$  and  $\Omega_0$ . They can be determined using the specifications of the bandpass analog filter using the following relations : –

$$\begin{aligned}\Omega_0 &= \sqrt{\Omega_{P_1}\Omega_{P_2}} = \sqrt{0.859 * 1.088} = 0.967 \\ B &= \Omega_{P_2} - \Omega_{P_1} = 1.088 - 0.859 = 0.229\end{aligned}$$

$\Omega$	$\Omega_L$
$0^+$	$-\infty$
$0.771(\Omega_{S_1})$	$-1.478(\Omega_{L_{S_1}})$
$0.859(\Omega_{P_1})$	$-1(\Omega_{L_{P_1}})$
$0.967(\Omega_0)$	$0$
$1.088(\Omega_{P_2})$	$1(\Omega_{L_{P_2}})$
$1.209(\Omega_{S_2})$	$1.902(\Omega_{L_{S_2}})$
$\infty$	$\infty$

## 1.5 Frequency Transformed Lowpass Analog Filter Specifications

- **Passband Edge** :  $1(\Omega_{L_P})$
- **Stopband Edge** :  $\min(-(\Omega_{L_{S_1}}), (\Omega_{L_{S_2}})) = 1.478(\Omega_{L_S})$
- **Tolerance** : 0.15 in magnitude for both Passband and Stopband
- **Passband Nature** : Monotonic
- **Stopband Nature** : Monotonic

## 1.6 Analog Lowpass Transfer Function

We need an Analog Filter which has a monotonic passband and a monotonic stopband. Therefore we need to design using the **Butterworth** approximation. Since the tolerance( $\delta$ ) in both passband and stopband is 0.15, we define two new quantities in the following way : –

$$D_1 = \frac{1}{(1 - \delta)^2} - 1 = \frac{1}{0.85^2} - 1 = 0.384$$

$$D_2 = \frac{1}{\delta^2} - 1 = \frac{1}{0.15^2} - 1 = 43.444$$

Now using the inequality on the order N of the filter for the Butterworth Approximation we get : –

$$N_{min} = \left\lceil \frac{\log \sqrt{\frac{D_2}{D_1}}}{\log \frac{\Omega_S}{\Omega_P}} \right\rceil = \lceil 6.05 \rceil = 7$$

The cut-off frequency ( $\Omega_c$ ) of the Analog LPF should satisfy the following constraint : –

$$\frac{\Omega_P}{D_1^{\frac{1}{2N}}} \leq \Omega_c \leq \frac{\Omega_S}{D_2^{\frac{1}{2N}}}$$

$$1.071 \leq \Omega_c \leq 1.129$$

Thus we can choose the value of  $\Omega_c$  to be 1.1

Now, the poles of the transfer function can be obtained by solving the equation : –

$$1 + \left(\frac{s}{j\Omega_c}\right)^{2N} = 1 + \left(\frac{s}{j1.1}\right)^{14} = 0$$

Solving for the roots of the above equation, we get : –

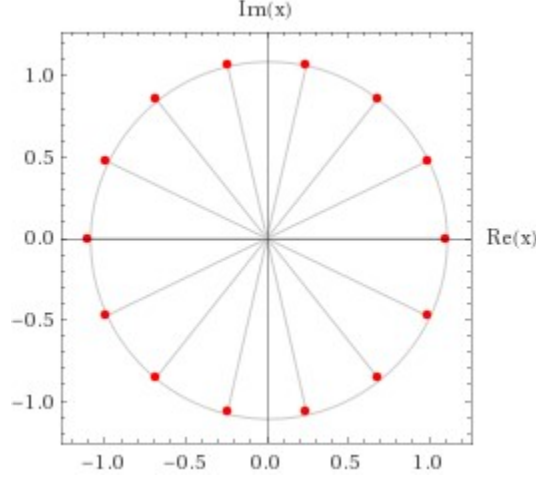


Figure 1: Poles of Magnitude Plot of Analog LPF

These are the Poles of the Magnitude Plot of the Transfer Function. In order to get a stable Analog LPF, we must include the poles lying in the Left Half Plane in the Transfer Function(The poles are symmetric about origin and we can pick one from each pair to be a part of our Transfer Function).

$$\begin{aligned} p_1 &= -1.1 \\ p_2 &= -0.991066 - j0.477272 \\ p_3 &= -0.991066 + j0.477272 \\ p_4 &= -0.685839 - j0.860015 \\ p_5 &= -0.685839 + j0.860015 \\ p_6 &= -0.244773 - j1.072420 \\ p_7 &= -0.244773 + j1.072420 \end{aligned}$$

Using the above poles which are in the left half plane we can write the Analog Lowpass Transfer Function as : –

$$\begin{aligned} H_{analog,LPF}(s_L) &= \frac{(\Omega_c)^N}{(s_L - p_1)(s_L - p_2)(s_L - p_3)(s_L - p_4)(s_L - p_5)(s_L - p_6)(s_L - p_7)} \\ &= \frac{1.949}{s_L^7 + 4.943s_L^6 + 12.218s_L^5 + 19.422s_L^4 + 21.364s_L^3 + 16.263s_L^2 + 7.961s_L + 1.949} \end{aligned}$$

Note that the scaling of the numerator is done in order to obtain a DC gain of 1.

## 1.7 Analog Bandpass Transfer Function

The transformation equation is given by : –

$$s_L = \frac{s^2 + \Omega_0^2}{Bs}$$

Substituting the values of the parameters  $B(0.229)$  and  $\Omega_0(0.967)$ , we get : –

$$s_L = \frac{s^2 + 0.935}{0.229s}$$

Substituting this value into  $H_{analog,LPF}(s_L)$  we get  $H_{analog,BPF}(s)$  as : –

$$\frac{6.411 \times 10^{-5} \cdot s^7}{\left( \frac{s^{14} + 1.131s^{13} + 7.176s^{12} + 6.572s^{11} + 21.356s^{10} + 15.676s^9 + 34.239s^8 + 19.659s^7 + 31.971s^6 + 13.668s^5 + 17.386s^4 + 4.995s^3 + 5.094s^2 + 0.75s + 0.619}{s^7} \right)}$$

## 1.8 Discrete Time Filter Transfer Function

To transform the analog domain transfer function into the discrete domain, we need to make use of the Bilinear Transformation which is given as : –

$$s = \frac{1 - z^{-1}}{1 + z^{-1}}$$

Using above equation we get  $H_{discrete,BPF}(z)$  from  $H_{analog,BPF}(s)(s)$  as : –

$$\frac{(3.6 \times 10^{-7}) \cdot (1 - 7z^{-2} + 21z^{-4} - 35z^{-6} + 35z^{-8} - 21z^{-10} + 7z^{-12} - z^{-14})}{\left( \frac{1 - 0.441z^{-1} + 5.92z^{-2} - 2.215z^{-3} + 15.032z^{-4} - 4.665z^{-5} + 21.216z^{-6} - 5.259z^{-7} + 17.968z^{-8} - 3.345z^{-9} + 9.129z^{-10} - 1.138z^{-11} + 2.575z^{-12} - 0.162z^{-13} + 0.311z^{-14}}{z^7} \right)}$$

## 1.9 Realization Using Direct Form II

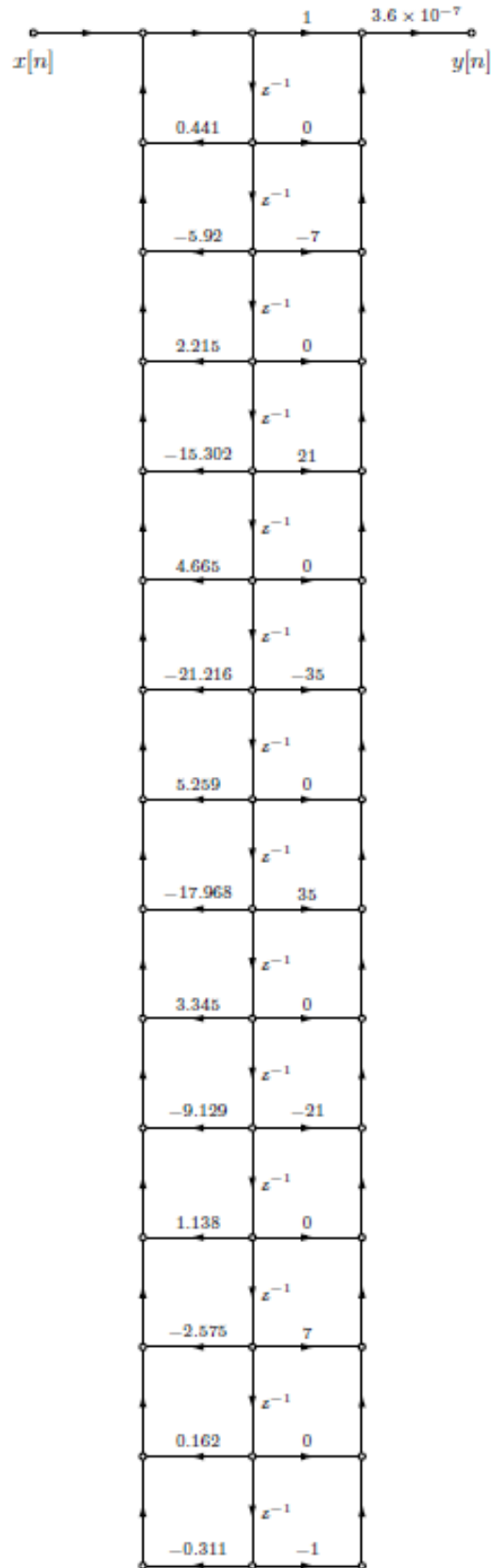


Figure 2: Direct Form II Block Diagram for  $H_{discrete,BPF}(z)$



Direct Form II is obtained by treating the transfer function  $H(z) = B(z)/(1 - A(z))$  as a cascade of  $1/(1 - A(z))$  followed by  $B(z)$ . The intermediate signal formed is the one whose samples get stored in the buffer. Thus, it is advantageous in comparison to Direct Form I since it saves memory space.

The negative of the denominator coefficients appear as gains on the side of the input sequence  $x[n]$  while the numerator coefficients appear on the side of the output  $y[n]$  as gains in the signal-flow graph representation of the Direct Form II.

## 1.10 FIR Filter Transfer Function using Kaiser Window

The tolerance in both the stopband and passband is given to be 0.15. Therefore,  $\delta = 0.15$  and we get the minimum stopband attenuation to be : –

$$A = -20\log(0.15) = 16.4728 \text{ dB}$$

Since  $A < 21$  dB, we get  $\beta$  to be 0 where  $\beta$  is the shape parameter of Kaiser window. Now to estimate the window length required, we use the empirical formula for the lower bound on the window length.

$$N \geq \frac{A - 7.95}{2 * 2.285 * \Delta\omega_T}$$

Here  $\Delta\omega_T$  is the minimum transition width. In our case, the transition width is the same on either side of the passband. With simulation, the tolerance wasn't satisfied using 20kHz. Hence, 10kHz was chosen as the transition width.

$$\Delta\omega_T = \frac{10\text{kHz} * 2\pi}{1.2\text{MHz}} = 0.066\pi$$

$$\therefore N \geq 29$$

The above equation gives a loose bound on the window length when the tolerance is not very stringent. On successive trials in MATLAB, it was found that a window length of **53** is required to satisfy the required constraints. Also, since  $\beta$  is 0, the window is actually a rectangular window.

The time domain coefficients were obtained by first generating the ideal impulse response samples for the same length as that of the window. The Kaiser Window was generated using the MATLAB function and applied on the ideal impulse response samples. For generating the ideal impulse response a separate function was made to generate the impulse response of Low-Pass filter. It took the cutoff value and the number of samples as input argument. The band-pass impulse response samples were generated as the difference between two low-pass filters as done in class.

The z-transform can simply be read off from the sequence values since its finite sequence.

## 2 Filter-2 (Bandstop) Details

### 2.1 Un-normalised discrete time filter specifications

Since Filter number is  $< 75$ ,  $m = 18$  and passband will be equiripple

$$q(m) = \lfloor 0.1m \rfloor = 1$$

$$r(m) = m - 10 \cdot q(m) = 18 - 10 = 8$$

$$B_L(m) = 157 + 17 \cdot q(m) + 13 \cdot r(m) = 278$$

$$B_H(m) = B_L(m) + 55 = 333$$

The second filter is given to be a **Band-stop** filter with stopband from  $B_L(m)$  kHz to  $B_H(m)$  kHz. Therefore the specifications are : –

- **Stopband** : 278 kHz to 333 kHz
- **Transition Band** : 20 kHz on either side of stopband
- **Passband** : 0-258 kHz and 353-600 kHz ( $\because$  Sampling rate is 1.2 MHz)
- **Tolerance** : 0.15 in magnitude for both Passband and Stopband
- **Passband Nature** : Equiripple
- **Stopband Nature** : Monotonic

### 2.2 Normalized Digital Filter Specifications

Sampling Rate = 1.2 MHz

In the normalized frequency axis, sampling rate corresponds to  $2\pi$

Thus, any frequency( $\Omega$ ) upto 600 kHz ( $= \frac{\text{SamplingRate}}{2}$ ) can be represented on the normalized axis( $\omega$ ) as : –

$$\omega = \frac{\Omega * 2\pi}{\text{SamplingRate}}$$

Therefore, the corresponding normalized digital filter specifications are : –

- **Stopband** :  $0.4633\pi$  to  $0.555\pi$
- **Transition Band** :  $0.033\pi$  on either side of passband
- **Passband** :  $0-0.43\pi$  and  $0.5883\pi-\pi$
- **Tolerance** : 0.15 in magnitude for both Passband and Stopband
- **Passband Nature** : Equiripple
- **Stopband Nature** : Monotonic

## 2.3 Analog filter specifications using Bilinear Transformation

The bilinear transformation is given as : –

$$\Omega = \tan\left(\frac{\omega}{2}\right)$$

Applying the Bilinear transform to the frequencies at the band-edges, we get : –

$\omega$	$\Omega$
0	0
$0.43\pi$	0.801
$0.4633\pi$	0.891
$0.555\pi$	1.19
$0.5883\pi$	1.324
$\pi$	$\infty$

Therefore the corresponding analog filter specifications for the same type of analog filter using the bilinear transformation are : –

- **Stopband** :  $0.891(\Omega_{S_1})$  to  $1.19(\Omega_{S_2})$
- **Transition Band** :  $0.801$  to  $0.891$  and  $1.19$  to  $1.324$
- **Passband** :  $0$  to  $0.801(\Omega_{P_1})$  and  $1.324(\Omega_{P_2})$  to  $\infty$
- **Tolerance** :  $0.15$  in magnitude for both Passband and Stopband
- **Passband Nature** : Equiripple
- **Stopband Nature** : Monotonic

## 2.4 Frequency Transformation and Relevant Parameters

We need to transform a Band-Stop analog filter to a Lowpass analog filter. We require two parameters in such a case. We can make use of the Bandpass transformation which is given as : –

$$\Omega_L = \frac{B\Omega}{\Omega_0^2 - \Omega^2}$$

The two parameters in the above equation are  $B$  and  $\Omega_0$ . They can be determined using the specifications of the bandpass analog filter using the following relations : –

$$\begin{aligned}\Omega_0 &= \sqrt{\Omega_{P_1}\Omega_{P_2}} = \sqrt{0.801 * 1.324} = 1.03 \\ B &= \Omega_{P_2} - \Omega_{P_1} = 1.324 - 0.801 = 0.526\end{aligned}$$

$\Omega$	$\Omega_L$
$0^+$	$0^+$
$0.801(\Omega_{P_1})$	$1(\Omega_{LP_1})$
$0.891(\Omega_{S_1})$	$1.755(\Omega_{LS_1})$
$1.03(\Omega_0^-)$	$+\infty$
$1.03(\Omega_0^+)$	$-\infty$
$1.19(\Omega_{S_2})$	$-1.762(\Omega_{LS_2})$
$1.324(\Omega_{P_2})$	$-1(\Omega_{LP_2})$
$\infty$	$0^-$

## 2.5 Frequency Transformed Lowpass Analog Filter Specifications

- **Passband Edge** :  $1(\Omega_{LP})$
- **Stopband Edge** :  $\min((\Omega_{LS_1}), -(\Omega_{LS_2})) = 1.755(\Omega_{LS})$
- **Tolerance** : 0.15 in magnitude for both Passband and Stopband
- **Passband Nature** : Equiripple
- **Stopband Nature** : Monotonic

## 2.6 Analog Lowpass Transfer Function

We need an Analog Filter which has a monotonic passband and a monotonic stopband. Therefore we need to design using the **Butterworth** approximation. Since the tolerance( $\delta$ ) in both passband and stopband is 0.15, we define two new quantities in the following way : –

$$D_1 = \frac{1}{(1 - \delta)^2} - 1 = \frac{1}{0.85^2} - 1 = 0.384$$

$$D_2 = \frac{1}{\delta^2} - 1 = \frac{1}{0.15^2} - 1 = 43.444$$

Now choosing the parameter  $\epsilon$  of the Chebyshev Filter to be  $\sqrt{D_1}$ , we get the min value of N as : –

$$N_{min} = \lceil \frac{\cosh^{-1}(\sqrt{\frac{D_2}{D_1}})}{\cosh^{-1}(\frac{\Omega_S}{\Omega_P})} \rceil = \lceil 2.63 \rceil = 3$$

Now, the poles of the transfer function can be obtained by solving the equation : –

$$1 + D_1 \cosh^2(N_{min} \cosh^{-1}(\frac{s}{j})) = 1 + 0.384 \cosh^2(3 \cosh^{-1}(\frac{s}{j})) = 0$$

Solving for the roots of the above equation, we get : –

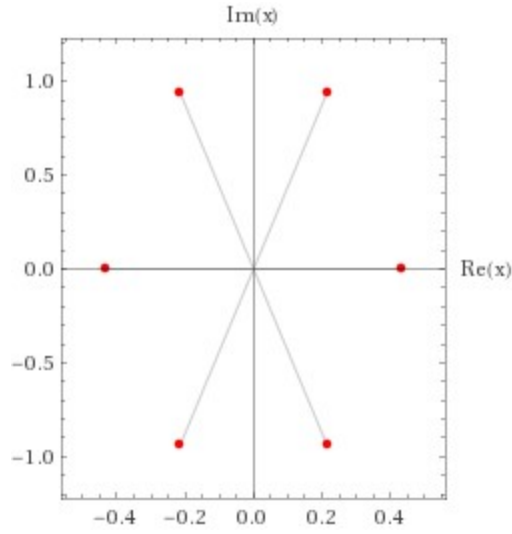


Figure 3: Poles of Magnitude Plot of Analog LPF

These are the Poles of the Magnitude Plot of the Transfer Function. In order to get a stable Analog LPF, we must include the poles lying in the Left Half Plane in the Transfer Function (The poles are symmetric about origin and we can pick one from each pair to be a part of our Transfer Function).

$$\begin{aligned} p_1 &= -0.43105 \\ p_2 &= -0.21553 + j0.94306 \\ p_3 &= -0.21553 - j0.94306 \end{aligned}$$

Using the above poles which are in the left half plane and the fact that N is odd we can write the Analog Lowpass Transfer Function as :- :-

$$\begin{aligned} H_{analog,LPF}(s_L) &= \frac{(-1)^3 p_1 p_2 p_3}{(s_L - p_1)(s_L - p_2)(s_L - p_3)} \\ &= \frac{0.407}{s_L^3 + 0.8621s_L^2 + 1.1216s_L + 0.4034} \end{aligned}$$

Note that since it is odd order, we take the DC Gain to be 1.

## 2.7 Analog Bandstop Transfer Function

The transformation equation is given by :-

$$s_L = \frac{Bs}{s^2 + \Omega_0^2}$$

Substituting the values of the parameters B(0.526) and  $\Omega_0$ (1.03), we get :-

$$s_L = \frac{0.526s}{s^2 + 1.061}$$

Substituting this value into  $H_{analog,LPF}(s_L)$  we get  $H_{analog,BSF}(s)$  as : –

$$\frac{s^6 + 3.184s^4 + 3.379s^2 + 1.195}{(s^6 + 1.456s^5 + 3.769s^4 + 3.445s^3 + 4s^2 + 1.639s + 1.195)}$$

The coefficients of odd powers of s in numerator are all 0.

## 2.8 Discrete Time Filter Transfer Function

To transform the analog domain transfer function into the discrete domain, we need to make use of the Bilinear Transformation which is given as : –

$$s = \frac{1 - z^{-1}}{1 + z^{-1}}$$

Using above equation we get  $H_{discrete,BSF}(z)$  from  $H_{analog,BSF}(s)(s)$  as : –

$$\frac{(0.531).(1 + 0.178z^{-1} + 3.011z^{-2} + 0.357z^{-3} + 3.011z^{-4} + 0.178z^{-5} + z^{-6})}{(1 + 0.143z^{-1} + 1.836z^{-2} + 0.181z^{-3} + 1.213z^{-4} + 0.054z^{-5} + 0.207z^{-6})}$$

## 2.9 Realization Using Direct Form II

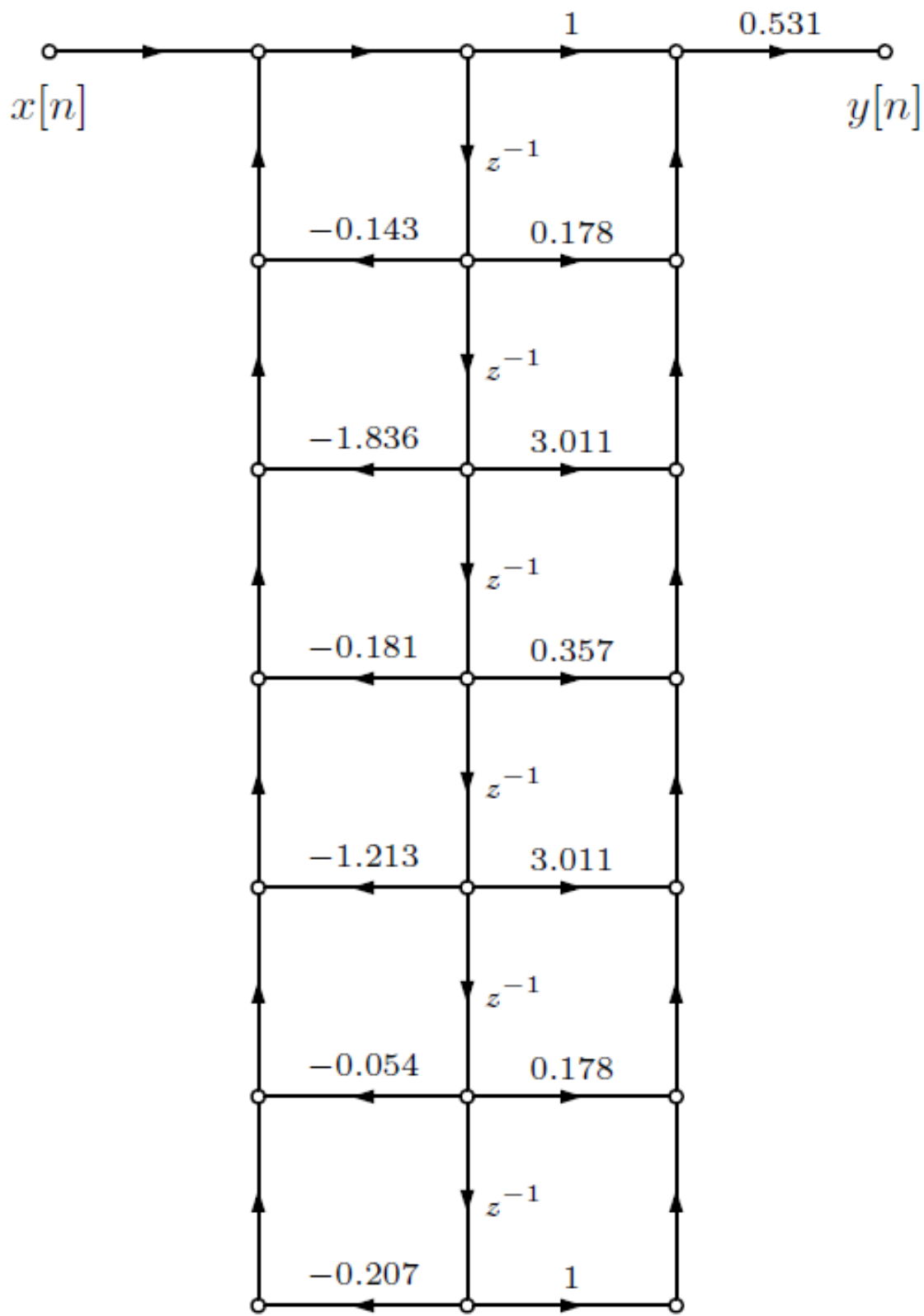


Figure 4: Direct Form II Block Diagram for  $H_{discrete,BSF}(z)$

Direct Form II is obtained by treating the transfer function  $H(z) = B(z)/(1 - A(z))$  as a cascade of  $1/(1 - A(z))$  followed by  $B(z)$ . The intermediate signal formed is the one whose samples get stored in the buffer. Thus, it is advantageous in comparison to Direct Form I since it saves memory space.

The negative of the denominator coefficients appear as gains on the side of the input sequence  $x[n]$  while the numerator coefficients appear on the side of the output  $y[n]$  as gains in the signal-flow graph representation of the Direct Form II.

## 2.10 FIR Filter Transfer Function using Kaiser Window

The tolerance in both the stopband and passband is given to be 0.15. Therefore,  $\delta = 0.15$  and we get the minimum stopband attenuation to be : –

$$A = -20\log(0.15) = 16.4728 \text{ dB}$$

Since  $A < 21$  dB, we get  $\beta$  to be 0 where  $\beta$  is the shape parameter of Kaiser window. Now to estimate the window length required, we use the empirical formula for the lower bound on the window length.

$$N \geq \frac{A - 7.95}{2 * 2.285 * \Delta\omega_T}$$

Here  $\Delta\omega_T$  is the minimum transition width. In our case, the transition width is the same on either side of the passband. With simulation, the tolerance wasn't satisfied using 20kHz. Hence, 10kHz was chosen as the transition width.

$$\Delta\omega_T = \frac{20kHz * 2\pi}{1.2MHz} = 0.066\pi$$

$$\therefore N \geq 29$$

The above equation gives a loose bound on the window length when the tolerance is not very stringent. On successive trials in MATLAB, it was found that a window length of **49** is required to satisfy the required constraints. Also, since  $\beta$  is 0, the window is actually a rectangular window.

The time domain coefficients were obtained by first generating the ideal impulse response samples for the same length as that of the window. The Kaiser Window was generated using the MATLAB function and applied on the ideal impulse response samples. For generating the ideal impulse response a separate function was made to generate the impulse response of Low-Pass filter. It took the cutoff value and the number of samples as input argument. The band-stop impulse response samples were generated as the difference between three low-pass filters( all-pass - bandpass ) as done in class.



## 3 MATLAB Plots

### 3.1 Filter 1 - Bandpass

#### 3.1.1 IIR Filter

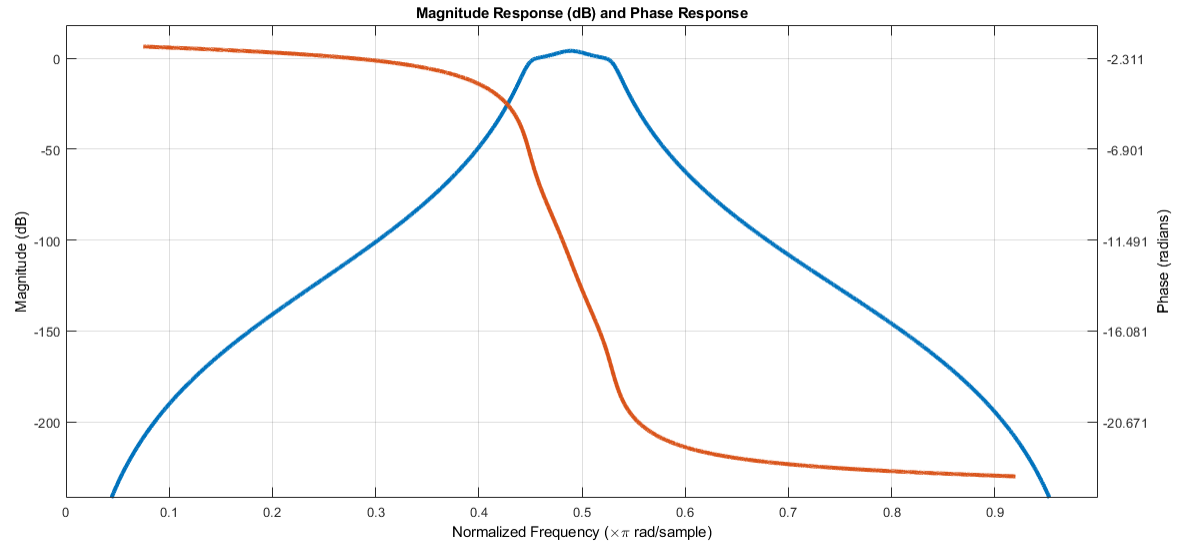


Figure 5: Frequency Response

From the above plot, it can be seen that the passband and stopband attenuation have been satisfied. It can be seen that the **phase response is not linear**

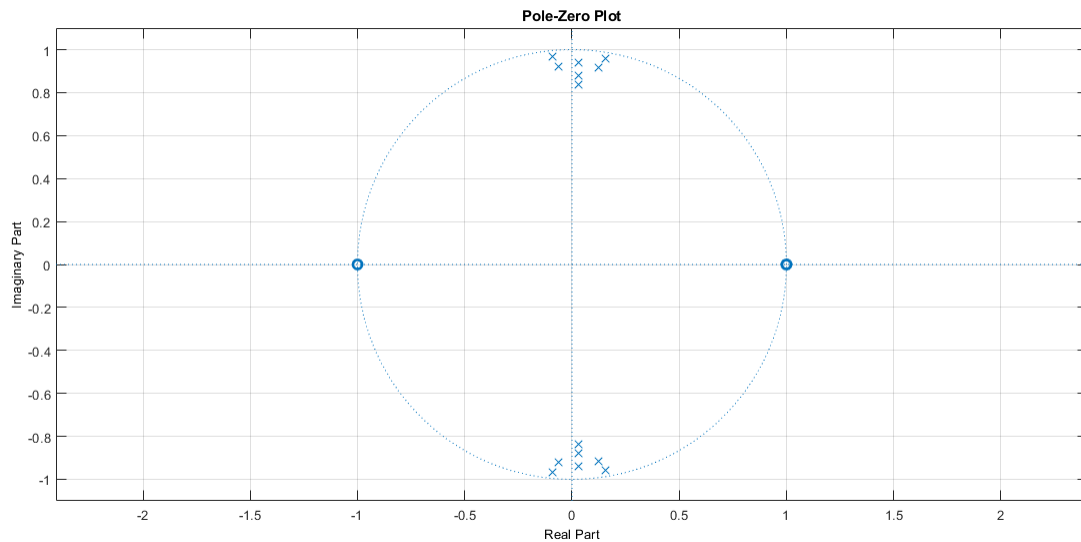


Figure 6: Pole-Zero map (all poles within unit circle, hence stable )

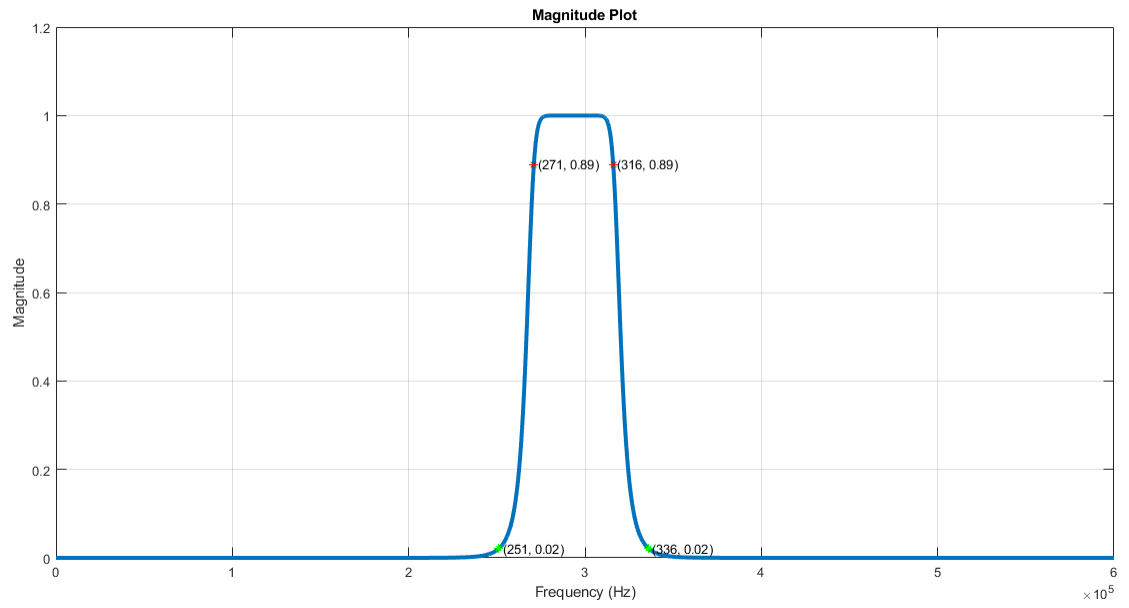


Figure 7: Magnitude Response

In the above plot, the band edge frequencies (in kHz) have been marked. From the magnitude at these frequencies it can be seen that the specifications required in the passband and the stopband have been met.

### 3.1.2 FIR Filter

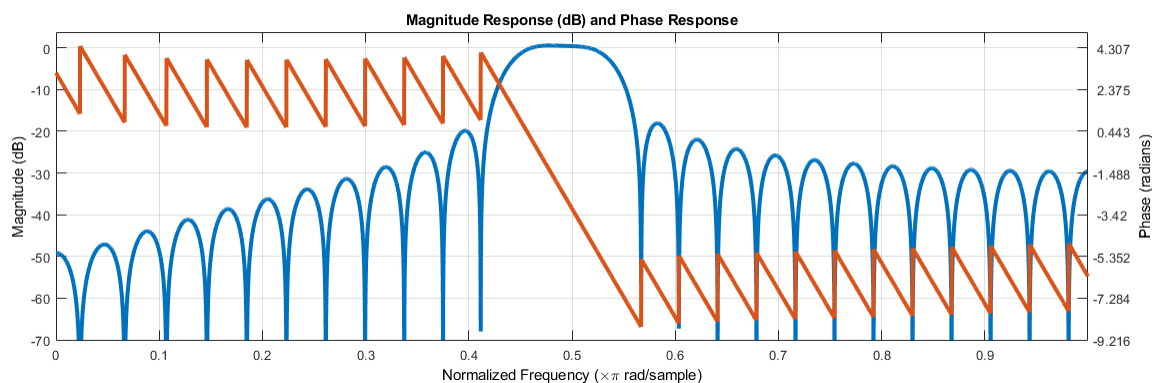


Figure 8: Frequency Response

From the above plot, the passband tolerance and stopband attenuation have been satisfied. It can be seen that the FIR Filter is indeed giving a **Linear Phase** response which is desired.

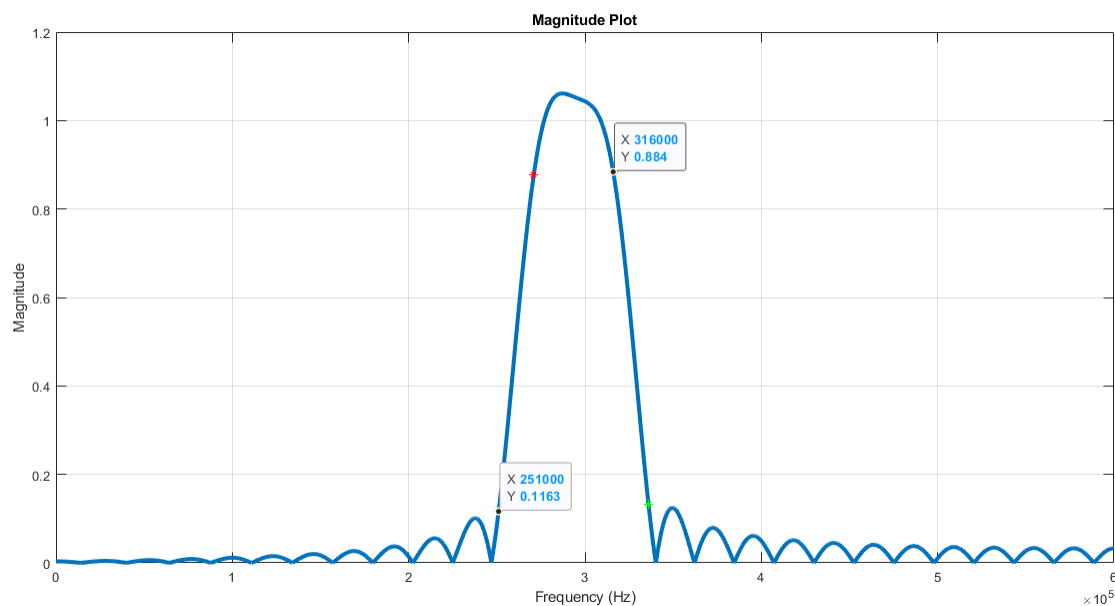


Figure 9: Magnitude Plot

In the above plot, the band edge frequencies have been marked. From the magnitude at these frequencies it can be seen that the specifications required in the passband and the stopband have been met.

## 3.2 Filter 2 - Bandstop

### 3.2.1 IIR Filter

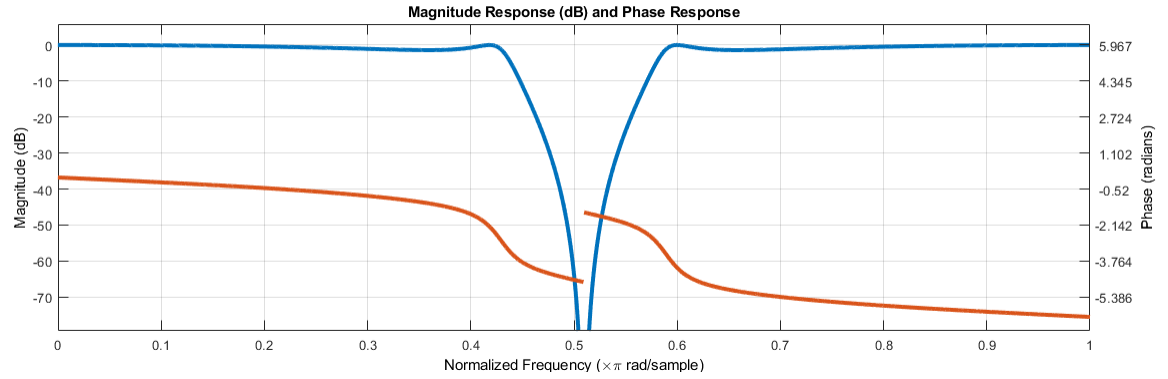


Figure 10: Frequency Response

From the above plot, it can be seen that the passband and stopband attenuation have been satisfied. It can be seen that the **phase response is not linear**

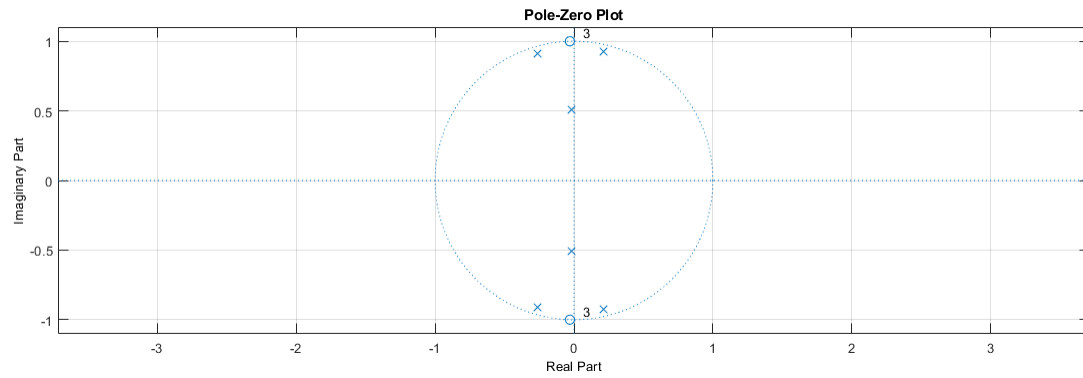


Figure 11: Pole-Zero map (all poles within unit circle, hence stable )

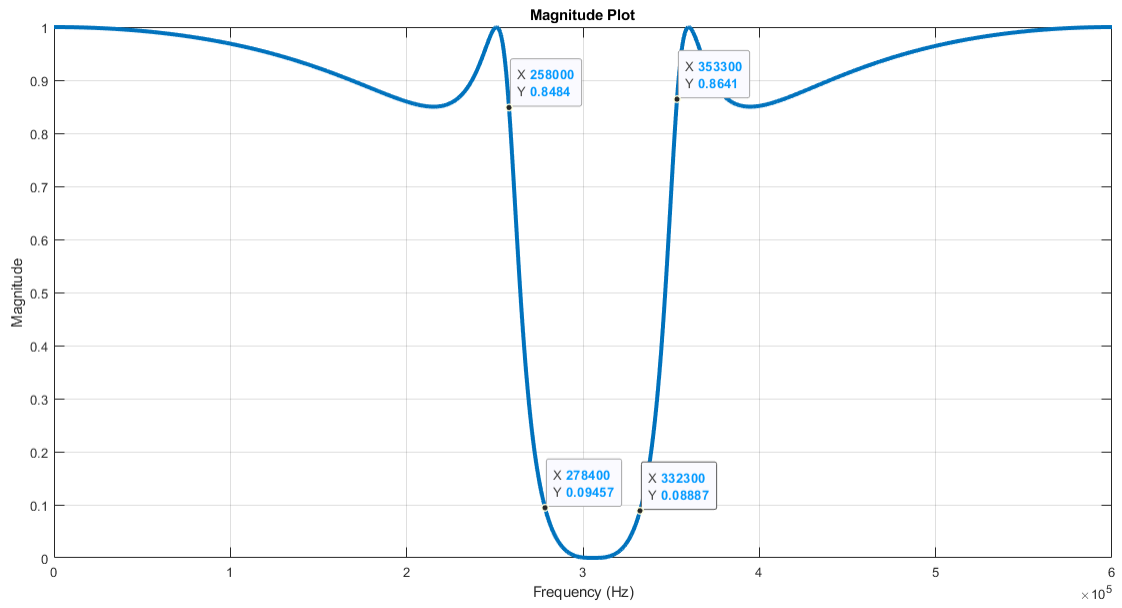


Figure 12: Magnitude Response

In the above plot, the band edge frequencies (in kHz) have been marked. From the magnitude at these frequencies it can be seen that the specifications required in the passband and the stopband have been met.

### 3.2.2 FIR Filter

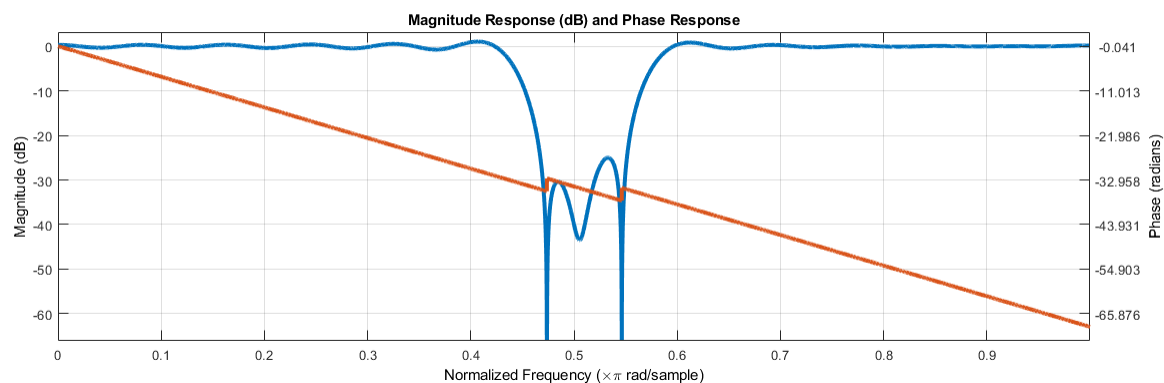


Figure 13: Frequency Response

From the above plot, the passband tolerance and stopband attenuation have been satisfied. It can be seen that the FIR Filter is indeed giving a **Linear Phase** response which is desired.

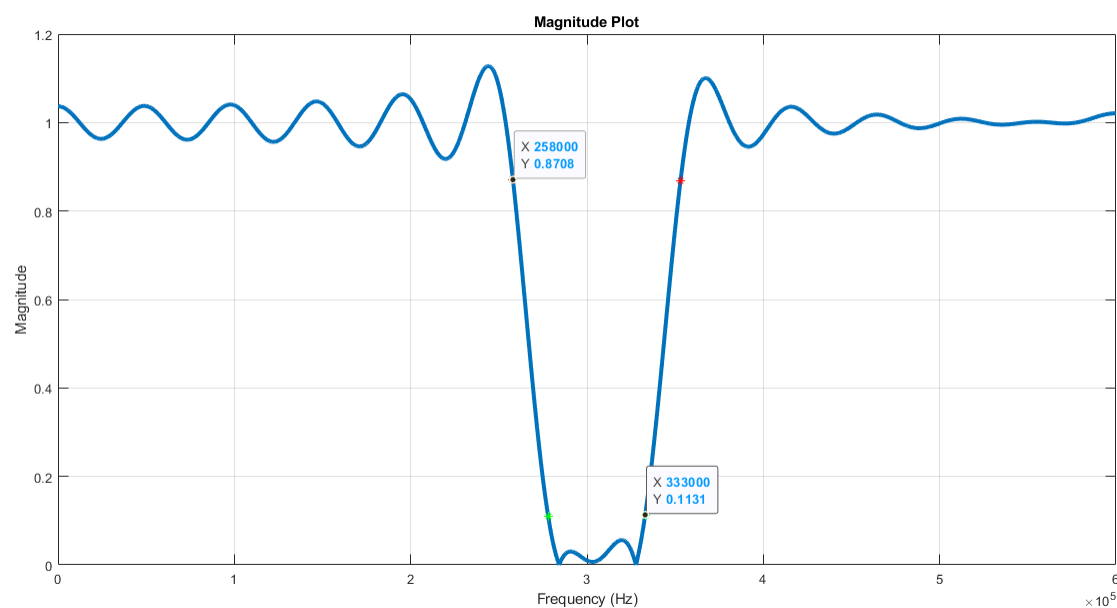


Figure 14: Magnitude Plot

In the above plot, the band edge frequencies have been marked. From the magnitude at these frequencies it can be seen that the specifications required in the passband and the stopband have been met.

## 4 Peer Review Acknowledgement

To whosoever it may concern,  
I have successfully reviewed this filter design report done by **Anugole Sai Gaurav** as a part of the course assignment for EE338, Digital Signal Processing . This includes the design process, context, calculations, simulation programs and results.

Thank you

Regards,  
**Jayesh Choudhary**  
**170070038**