# EE338 : FIR Filter Design

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## 1 Student Details

**Name :** Dharod Sahil Nilesh
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## 2 Bandpass Filter

### 2.1 Un-normalized Discrete Time Filter Specifications

Filter Number = 100
Since filter number > 80, m = 100 - 80 = 20 and passband will be equiripple.
q(m) = greatest integer strictly less than 0.1*m = 1
r(m) = m - 10*q(m) = 10
$B_L$(m) = 10 + 5*q(m) + 13*r(m) = 10 + 5*1 + 13*10 = 145KHz
$B_H$(m) = $B_L$(m) + 75 = 220KHz

The specifications of this filter are :

- Passband : **145 - 220 KHz**
- Transition band : **5KHz** on either side of passband
- Stopband : **0 - 140** and **225 - 300 KHz** (As **sampling rate** is **600KHz**)
- Tolerance : **0.15** in **magnitude** for both passband and stopband

### 2.2 Normalized Digital Filter Specifications

Sampling rate = 600KHz
In the normalized frequency axis, sampling rate corresponds to $2\pi$

Therefore, any frequency can be normalized as follows :

$$\omega = \frac{\Omega * 2\pi}{\Omega_s}$$

where $\Omega_s$ is the Sampling Rate.

For the normalized discrete filter specifications, the nature and tolerances being the dependent variables remain the same while the passband and stopband frequencies change as per the above transformations.

- Passband : **0.483 - 0.733** $\pi$
- Transition band : **0.0167** $\pi$ on either side of passband
- Stopband : **0 - 0.466** $\pi$ and **0.75 - 1** $\pi$

# 3 FIR Bandpass Filter

Both the passband and stopband tolerances are given to be 0.15
Therefore $\delta = 0.15$ and the minimum stopband attenuation A is given by :

$$A = -20log(\delta) = -20log(0.15) = 16.478$$

Since A < 21, we get $\beta = 0$, 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

$$2N_{min} + 1 \geq 1 + \frac{A - 7.95}{2.285 * \Delta\omega_T}$$

Here $\Delta\omega_T$ is the transition width which is the same on either side of the passband

$$\Delta\omega_T = \frac{5KHz * 2\pi}{600KHz} = 0.0167\pi$$

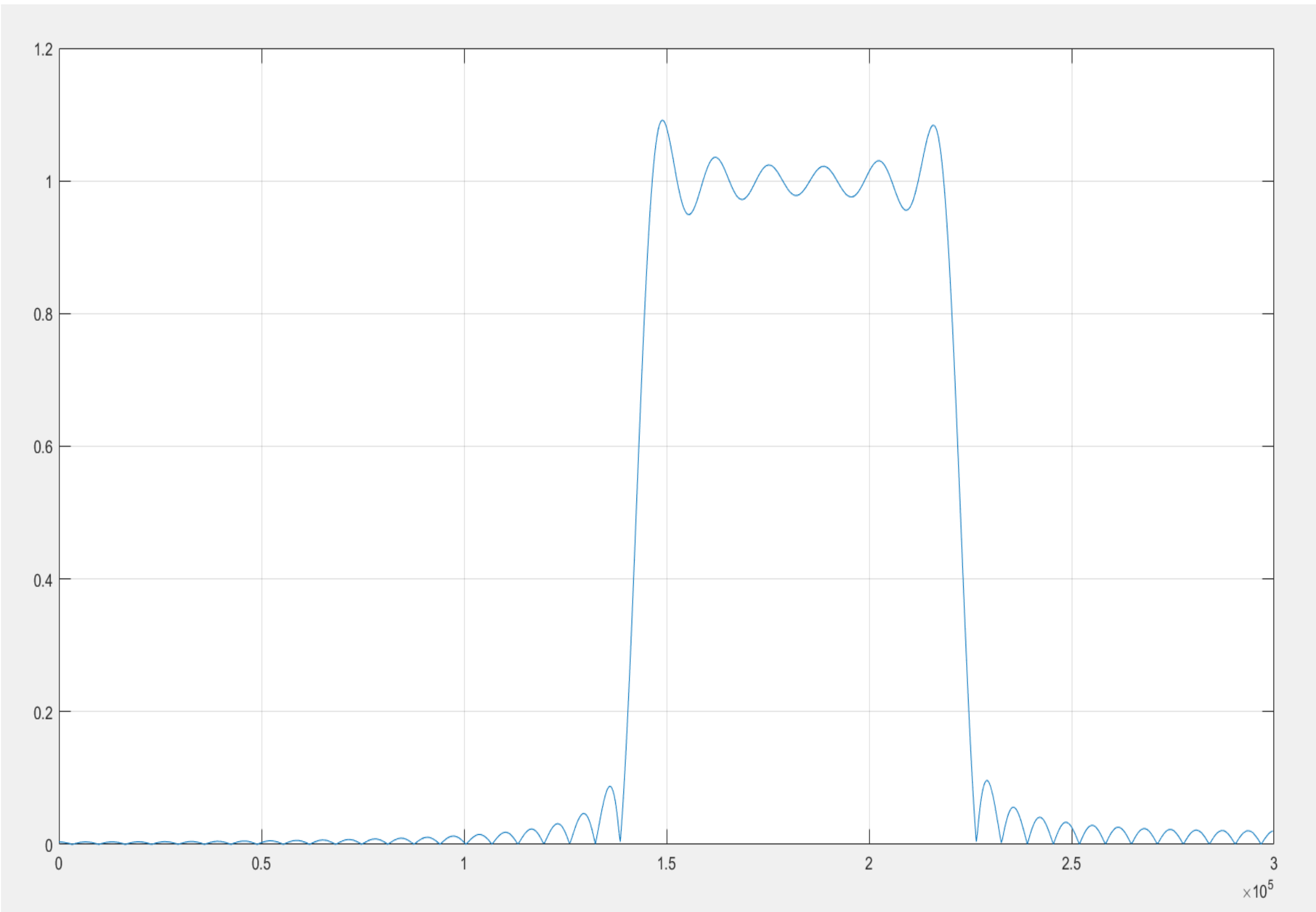
$$2N_{min} \geq 71.279$$

Hence we initially choose $N_{min} = 36$ ( $N_{min}$ is such that total number of samples is $2N_{min}$+1) Further for stringent tolerance and transition band specifications, we get $N_{total} = 2N_{min} + 23 = 95$ using trial and error.

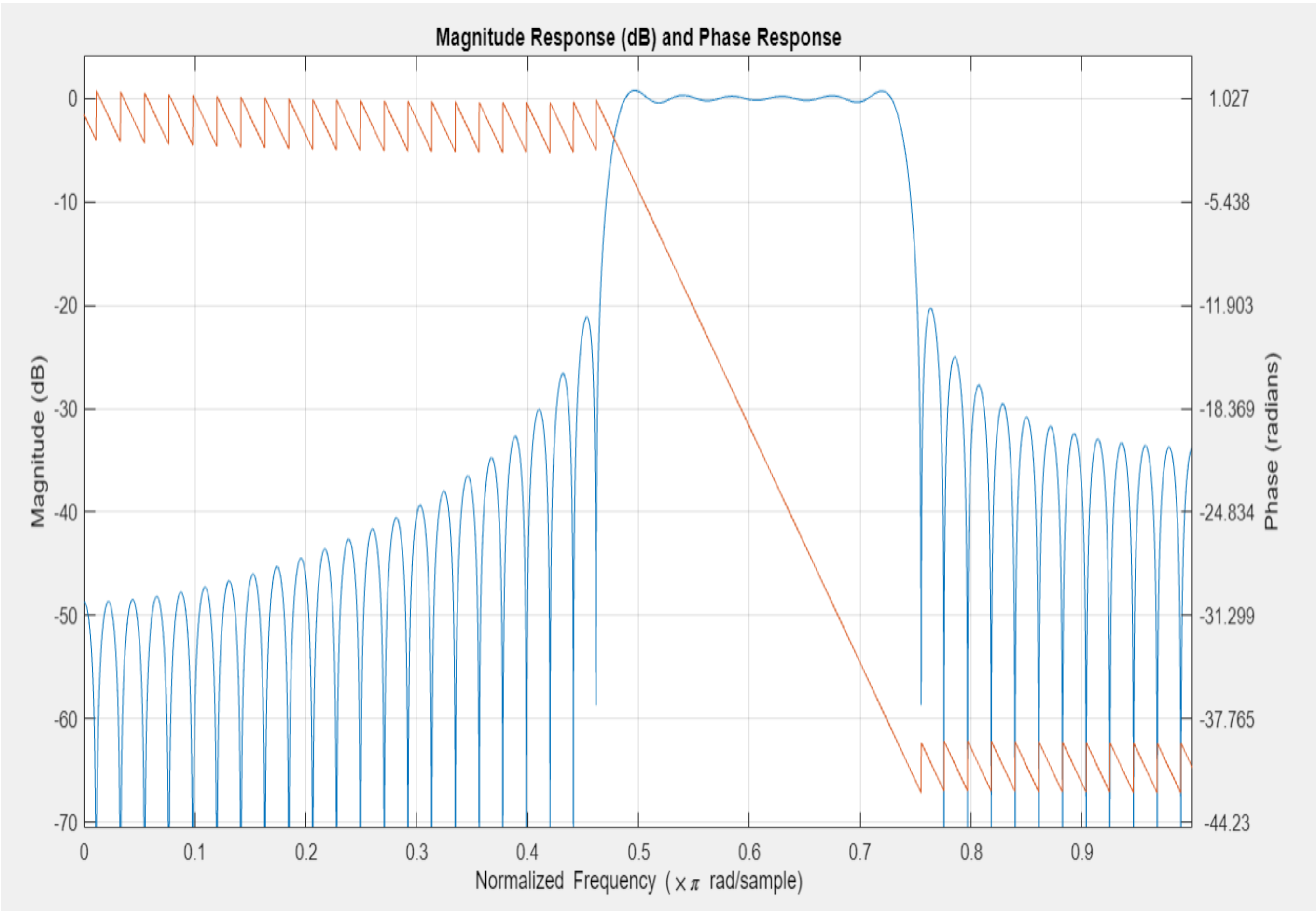
The time domain coefficients were obtained by first generating the ideal impulse response samples for the same length as that of the window. The Kaiser Window was generated using the MATLAB function and applied on the ideal impulse response samples. For generating the ideal impulse response a separate 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 with the cutoff frequencies being average of $\Omega_{s1}, \Omega_{p1}$ and $\Omega_{p2}, \Omega_{s2}$ respectively so that magnitude response reaches half of its peak value at the average of passband and stopband frequencies i.e. 0.475 $\pi$ and 0.7417 $\pi$

# 4 Matlab Plots

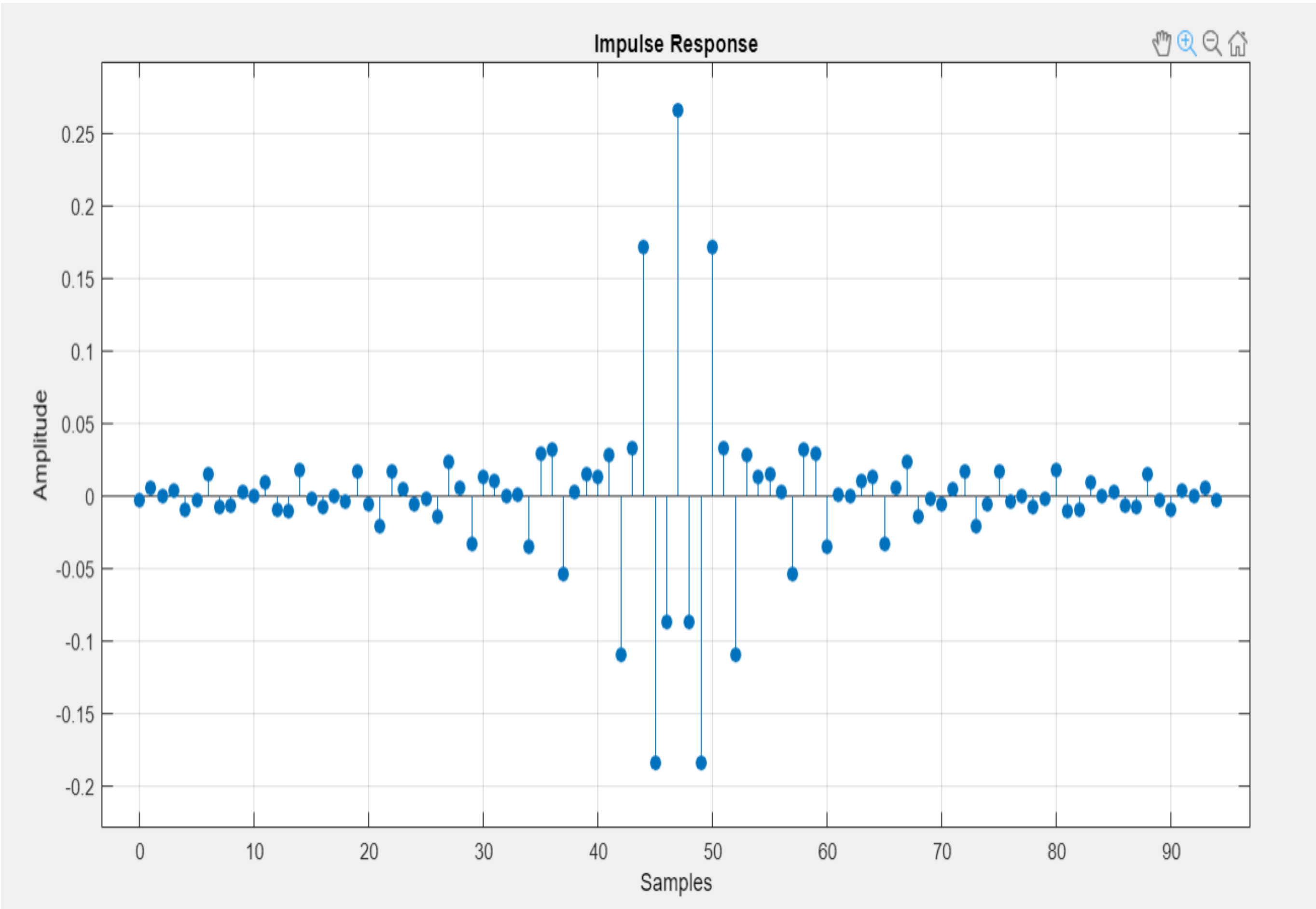
## 4.1 Frequency Response



## 4.2 Magnitude and Phase Response



### 4.3 Impulse Response



### 4.4 Coefficients

```
FIR_BandPass =

  Columns 1 through 9

   -0.0029    0.0057   -0.0000    0.0044   -0.0096   -0.0023    0.0152   -0.0069   -0.0063

  Columns 10 through 18

    0.0033   -0.0001    0.0099   -0.0096   -0.0102    0.0178   -0.0015   -0.0076    0.0000

  Columns 19 through 27

   -0.0038    0.0168   -0.0052   -0.0204    0.0175    0.0048   -0.0058   -0.0021   -0.0135

  Columns 28 through 36

    0.0238    0.0061   -0.0332    0.0133    0.0109   -0.0000    0.0010   -0.0349    0.0296

  Columns 37 through 45

    0.0326   -0.0533    0.0032    0.0151    0.0130    0.0283   -0.1093    0.0329    0.1721

  Columns 46 through 54

   -0.1838   -0.0865    0.2667   -0.0865   -0.1838    0.1721    0.0329   -0.1093    0.0283
```

```
Columns 55 through 63

   0.0130    0.0151    0.0032   -0.0533    0.0326    0.0296   -0.0349    0.0010   -0.0000

Columns 64 through 72

   0.0109    0.0133   -0.0332    0.0061    0.0238   -0.0135   -0.0021   -0.0058    0.0048

Columns 73 through 81

   0.0175   -0.0204   -0.0052    0.0168   -0.0038    0.0000   -0.0076   -0.0015    0.0178

Columns 82 through 90

  -0.0102   -0.0096    0.0099   -0.0001    0.0033   -0.0063   -0.0069    0.0152   -0.0023

Columns 91 through 95

  -0.0096    0.0044   -0.0000    0.0057   -0.0029
```

# 5 Bandstop Filter

## 5.1 Un-normalized Discrete Time Filter Specifications

Filter Number = 100
Since filter number > 80, m = 100 - 80 = 20 and passband will be monotonic.
q(m) = greatest integer strictly less than 0.1*m = 1
r(m) = m - 10*q(m) = 10
$B_L$(m) = 20 + 3*q(m) + 11*r(m) = 20 + 3*1 + 11*10 = 133KHz
$B_H$(m) = $B_L$(m) + 40 = 173KHz

The specifications of this filter are :

- Stopband : **133 - 173 KHz**
- Transition band : **5KHz** on either side of stopband
- Passband : **0 - 128** and **178 - 212.5 KHz** (As **sampling rate** is **425KHz**)
- Tolerance : **0.15** in **magnitude** for both passband and stopband
- Nature : Both passband and stopband are **monotonic**

## 5.2 Normalized Digital Filter Specifications

Sampling rate = 425KHz
In the normalized frequency axis, sampling rate corresponds to $2\pi$
Therefore, any frequency can be normalized as follows :

$$\omega = \frac{\Omega * 2\pi}{\Omega_s}$$

where $\Omega_s$ is the Sampling Rate.

For the normalized discrete filter specifications, the nature and tolerances being the dependent variables remain the same while the passband and stopband frequencies change as per the above transformations.

- Stopband : **0.626 - 0.814** $\pi$
- Transition band : **0.024** $\pi$ on either side of stopband
- Passband : **0 - 0.602** $\pi$ and **0.838 - 1** $\pi$

## 6 FIR Bandstop Filter

Both the passband and stopband tolerances are given to be 0.15
Therefore $\delta = 0.15$ and the minimum stopband attenuation A is given by :

$$A = -20log(\delta) = -20log(0.15) = 16.478$$

Since A $< 21$, we get $\beta = 0$, 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

$$2N_{min} + 1 \geq 1 + \frac{A - 7.95}{2.285 * \Delta\omega_T}$$

Here $\Delta\omega_T$ is the transition width which is the same on either side of the passband

$$\Delta\omega_T = \frac{5KHz * 2\pi}{425KHz} = 0.02353\pi$$

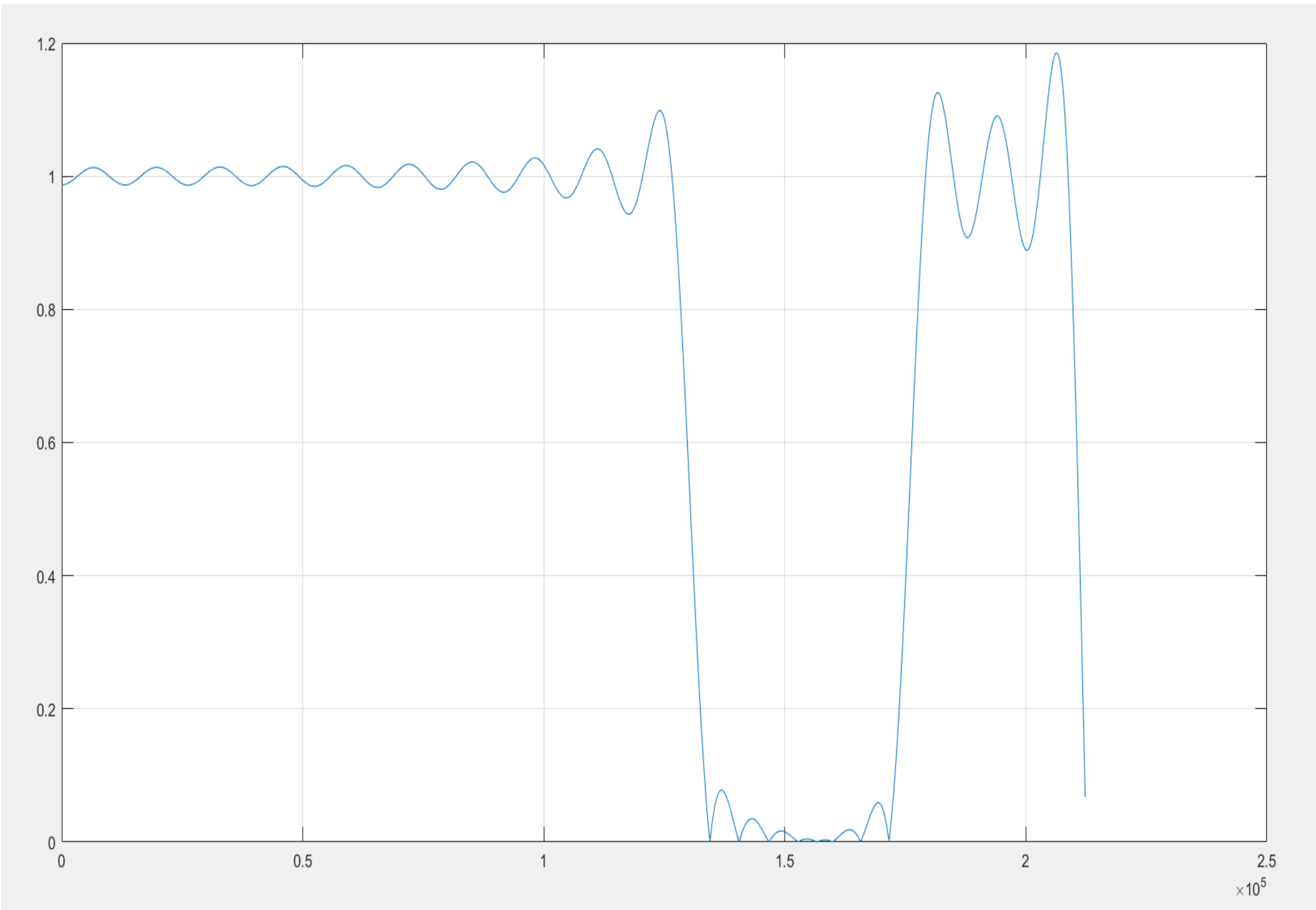
$$2N_{min} \geq 50.49$$

Hence we initially choose $N_{min} = 26$ ( $N_{min}$ is such that total number of samples is $2N_{min}$+1). Further for stringent tolerance and transition band specifications, we get $N_{total} = 2N_{min} + 13 = 65$ using trial and error.

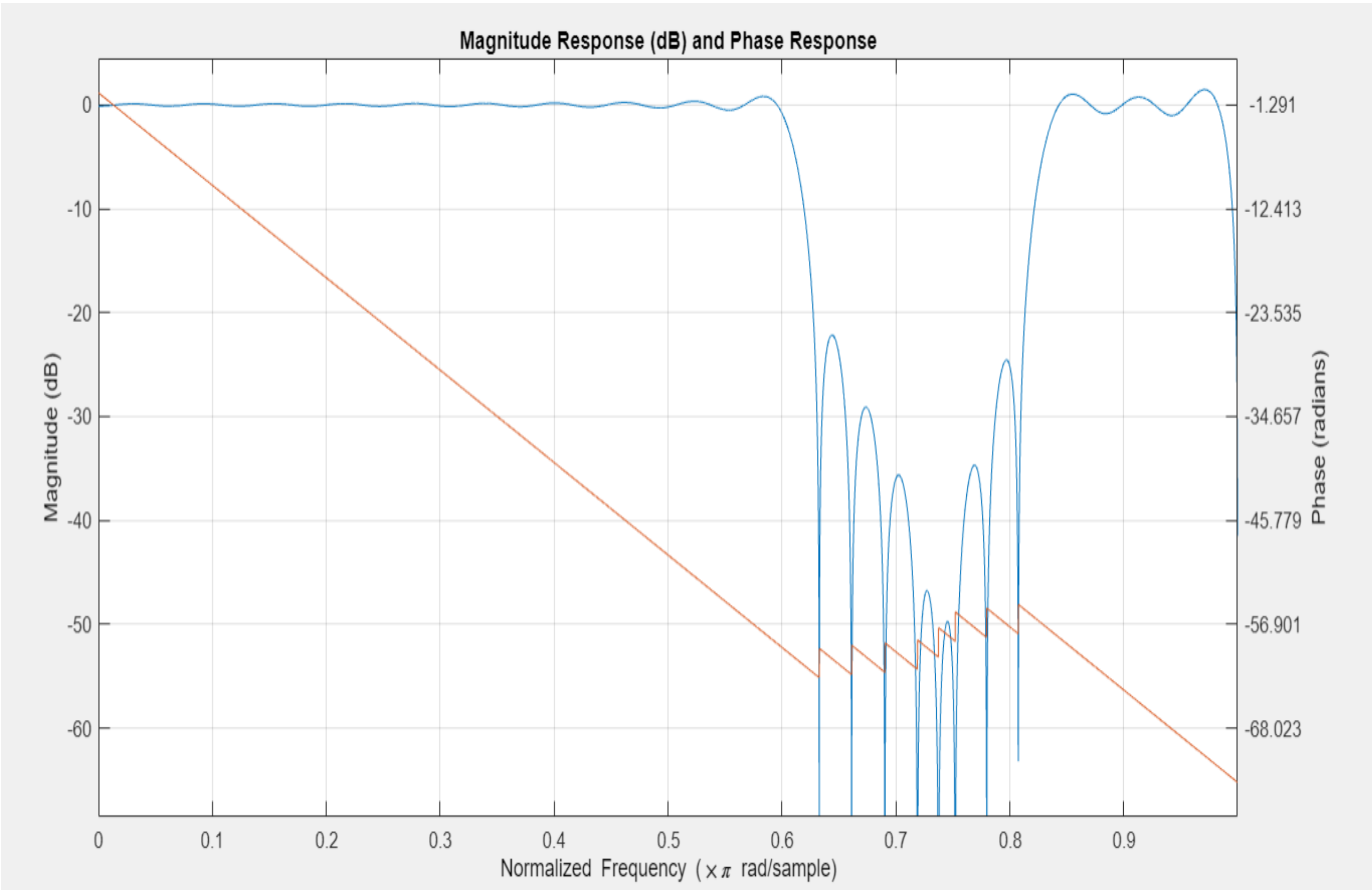
The time domain coefficients were obtained by first generating the ideal impulse response samples for the same length as that of the window. The Kaiser Window was generated using the MATLAB function and applied on the ideal impulse response samples. For generating the ideal impulse response a separate 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 an all pass filter and a band pass filter such that the cutoff frequencies are again at average of passband and stopband frequencies i.e. 0.614 $\pi$ and 0.826 $\pi$

# 7 Matlab Plots

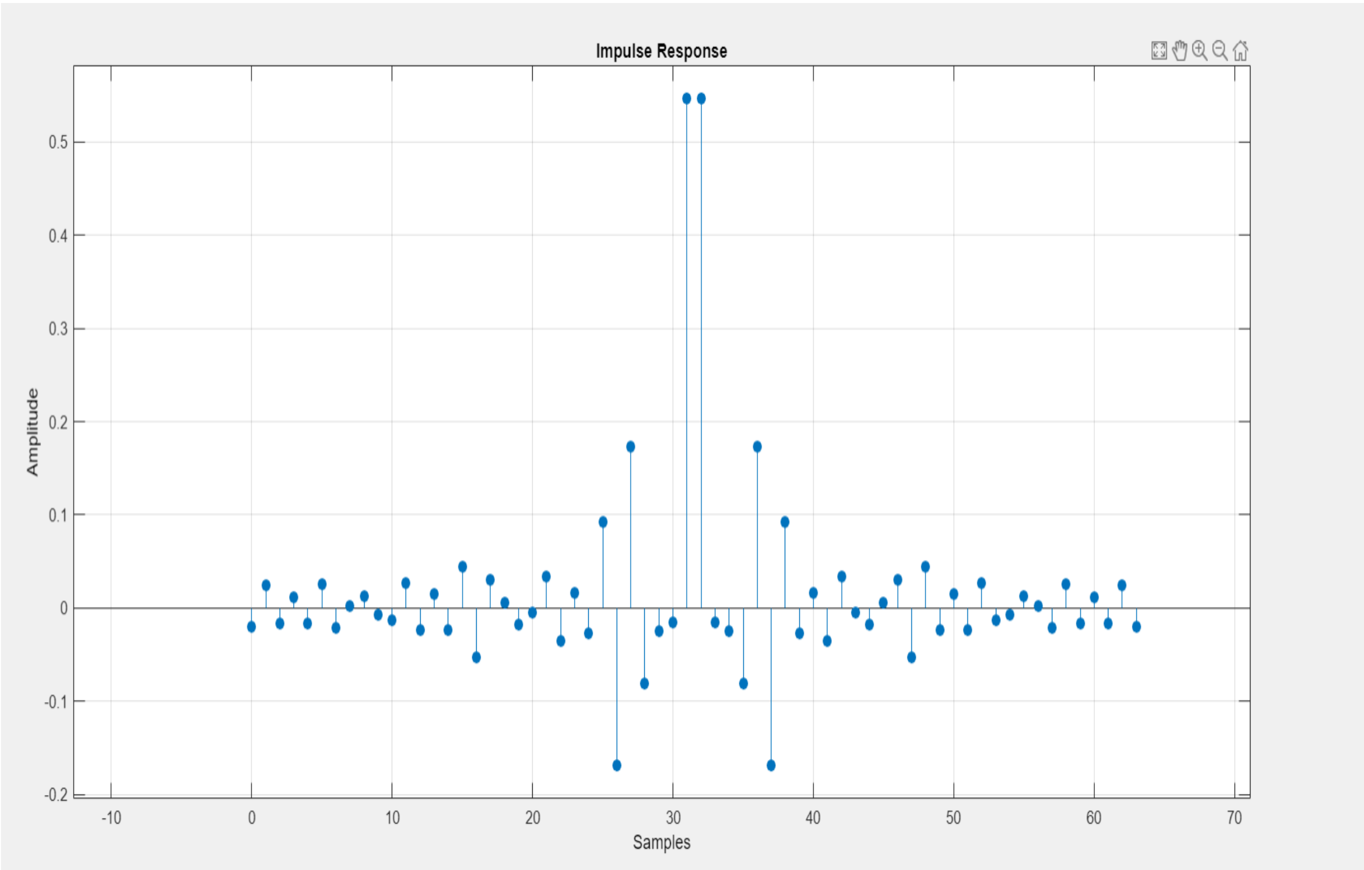
## 7.1 Frequency Response



### 7.2 Magnitude and Phase Response



## 7.3 Impulse Response



### 7.4 Coefficients

```
FIR_BandStop =

  Columns 1 through 9

   -0.0196    0.0243   -0.0169    0.0111   -0.0165    0.0252   -0.0210    0.0024    0.0127

  Columns 10 through 18

   -0.0073   -0.0134    0.0272   -0.0231    0.0150   -0.0231    0.0447   -0.0530    0.0301

  Columns 19 through 27

    0.0058   -0.0180   -0.0053    0.0343   -0.0348    0.0161   -0.0267    0.0925   -0.1688

  Columns 28 through 36

    0.1736   -0.0804   -0.0251   -0.0153    0.5468    0.5468   -0.0153   -0.0251   -0.0804

  Columns 37 through 45

    0.1736   -0.1688    0.0925   -0.0267    0.0161   -0.0348    0.0343   -0.0053   -0.0180
```

```
Columns 37 through 45

  0.1736   -0.1688    0.0925   -0.0267    0.0161   -0.0348    0.0343   -0.0053   -0.0180

Columns 46 through 54

  0.0058    0.0301   -0.0530    0.0447   -0.0231    0.0150   -0.0231    0.0272   -0.0134

Columns 55 through 63

 -0.0073    0.0127    0.0024   -0.0210    0.0252   -0.0165    0.0111   -0.0169    0.0243

Column 64

 -0.0196
```

## 8 Comparison between FIR and IIR Filters :

- FIR filters are easier to design as we only need to truncate the ideal impulse response using a suitable window function, instead of applying the bilinear and frequency transformation, designing a low-pass filter and then converting it to bandpass/bandstop as per our requirement.
- We get a linear (or psuedo linear) phase response in FIR Filter which we don't get in IIR Filters.
- We can't control the passband and stopband tolerances individually in FIR Filters, nor can we change their nature (monotonic or equiripple), which was possible in IIR Filters.
- We usually need a lot more resources for FIR filters as compared to IIR filters, as we can see that the value of N for FIR filters is considerably large.

## 9 Review

- I have verified the filter design of my team-mate Siddharth Kaushik (Roll No : 210070086)