INDIAN INSTITUTE OF TECHNOLOGY, BOMBAY

EE338 - DIGITAL SIGNAL PROCESSING

Digital Filter design

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1 FIR filter transfer function using Kaiser Window

1.1 Bandpass Filter

The tolerance for both the stopband and passband is given to be $\delta = 0.15$ and we get the minimum stopband attenuation as

$$A = -20log_{10}(\delta) = -16.4782dB \tag{1}$$

Since A ≤ 21 we get β to be 0 where β is the shape parameter of the Kaiser window. In order to estimate the required window length, we use the empirical formula for the lower bound on the window length.

$$N_{min} = \frac{A-8}{2.285 \times 2 \times \Delta\omega_T} \tag{2}$$

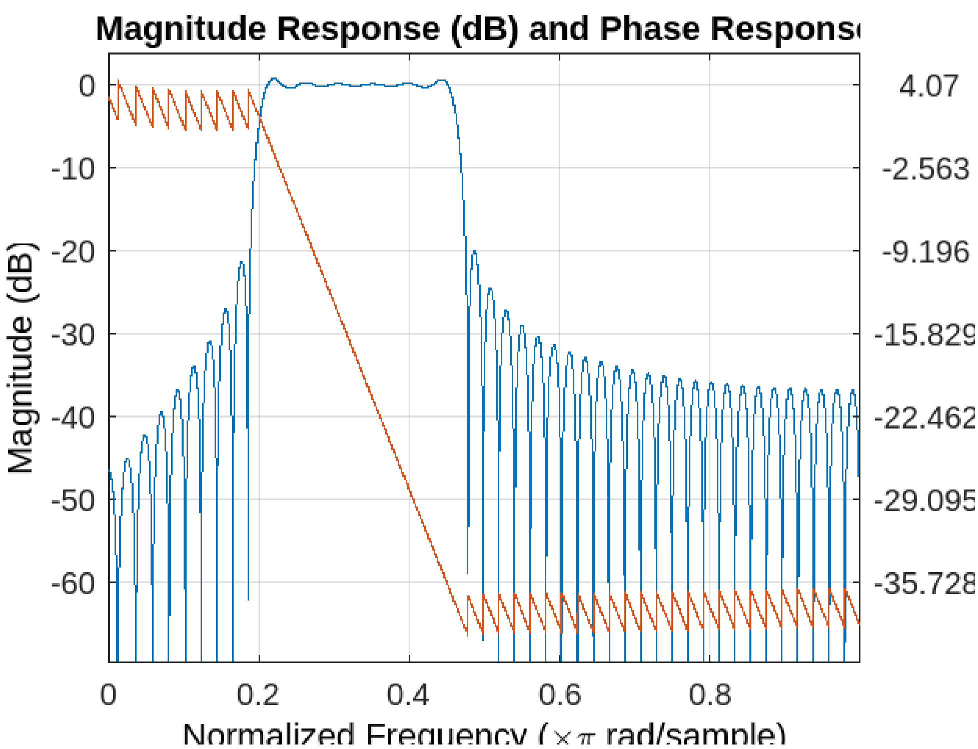
Here ω_T is the minimum transition width from the stopband to the passband.

$$\Delta\omega_T = \frac{5kHz \times 2 \times \pi}{600kHz} = 0.0167\pi \tag{3}$$

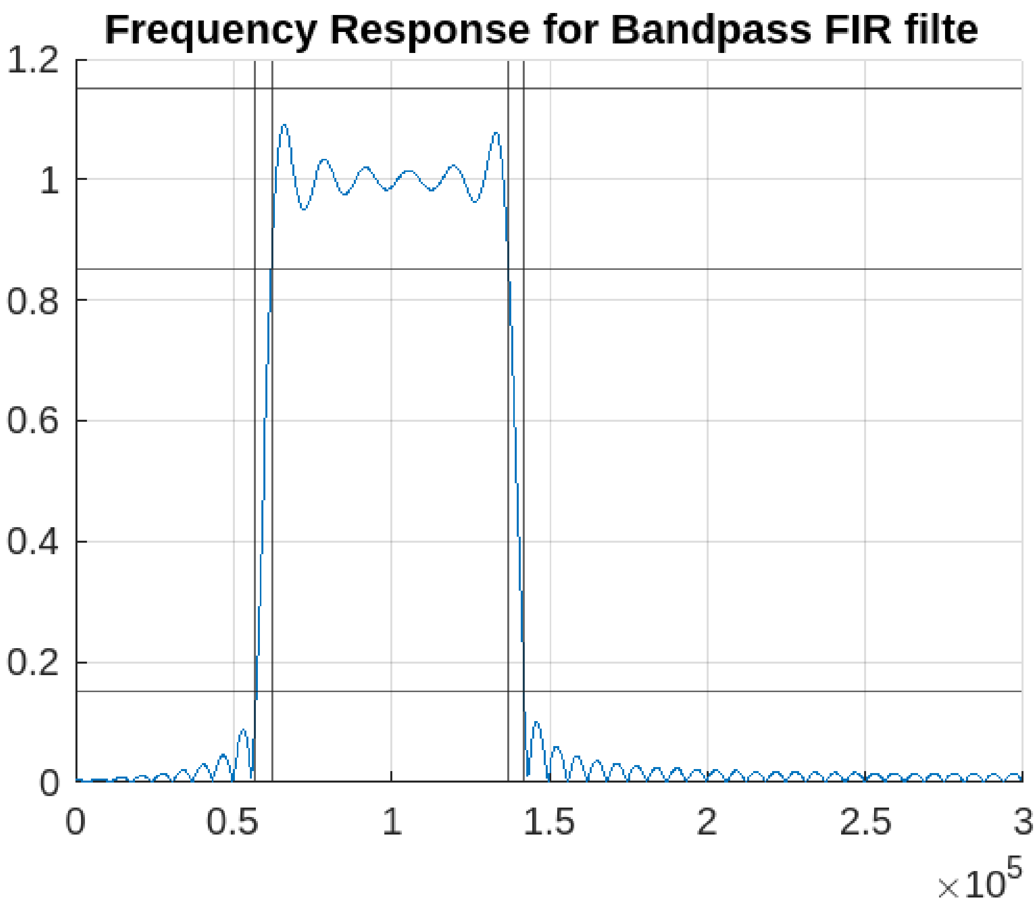
Substituting $\Delta\omega_T$ we get $N_{min} = 71$

The above equation gives a loose bound on the window length when the tolerance in not very stringent. From trial and error in MATLAB and checking if the specific given for the filter that I had to design the window length after this was **96**. The window is a rectangular window as $\beta = 0$.

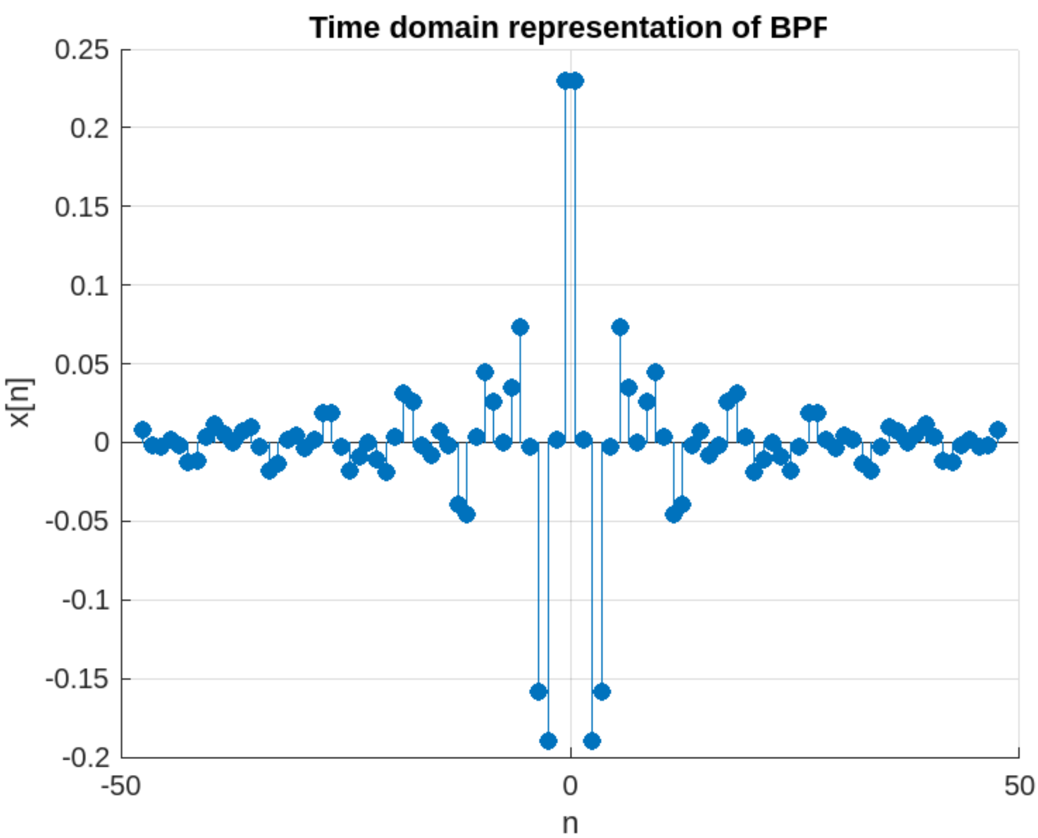
The time domain coefficients were obtained by first generating the ideal impulse response samples for the same length as that of the window. The Kaiser window was generated using the MATLAB function and applied on the ideal impulse response. The ideal impulse response was generated using a linear combination of impulse response of the ideal lowpass filter. The cutoff value and number of samples were given as the input argument to the ideal lowpass filter generating function.



Mynd 1: Magnitude Phase Response



Mynd 2: Frequnecy Response



Mynd 3: Impulse Response

```
Columns 1 through 13

   0.0083   -0.0019   -0.0028    0.0022   -0.0019   -0.0124   -0.0113    0.0031    0.0114    0.0050    0.0000    0.0067    0.0101

  Columns 14 through 26

  -0.0032   -0.0178   -0.0130    0.0020    0.0041   -0.0035    0.0020    0.0186    0.0188   -0.0032   -0.0178   -0.0088   -0.0000

  Columns 27 through 39

  -0.0110   -0.0186    0.0032    0.0312    0.0257   -0.0020   -0.0077    0.0075   -0.0020   -0.0396   -0.0459    0.0032    0.0443

  Columns 40 through 52

   0.0257         0    0.0352    0.0732   -0.0032   -0.1583   -0.1895    0.0020    0.2296    0.2296    0.0020   -0.1895   -0.1583

  Columns 53 through 65

  -0.0032    0.0732    0.0352         0    0.0257    0.0443    0.0032   -0.0459   -0.0396   -0.0020    0.0075   -0.0077   -0.0020

  Columns 66 through 78

   0.0257    0.0312    0.0032   -0.0186   -0.0110   -0.0000   -0.0088   -0.0178   -0.0032    0.0188    0.0186    0.0020   -0.0035

  Columns 79 through 91

   0.0041    0.0020   -0.0130   -0.0178   -0.0032    0.0101    0.0067    0.0000    0.0050    0.0114    0.0031   -0.0113   -0.0124

  Columns 92 through 96

  -0.0019    0.0022   -0.0028   -0.0019    0.0083
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Mynd 4: Coefficients for discrete impulse response

1.2 Bandstop

Since we have similar tolerance requirements as we had for bandstop. This means that we will get similar results for A and its corresponding β value.

$$A = -20log_{10}(\delta) = -16.4782dB \tag{4}$$

Since A $\leq$ 21 we get β to be 0 where β is the shape parameter of the Kaiser window. In order to estimate the required window length, we use the empirical formula for the lower bound on the window length.

$$N_{min} = \frac{A-8}{2.285 \times 2 \times \Delta\omega_T} \tag{5}$$

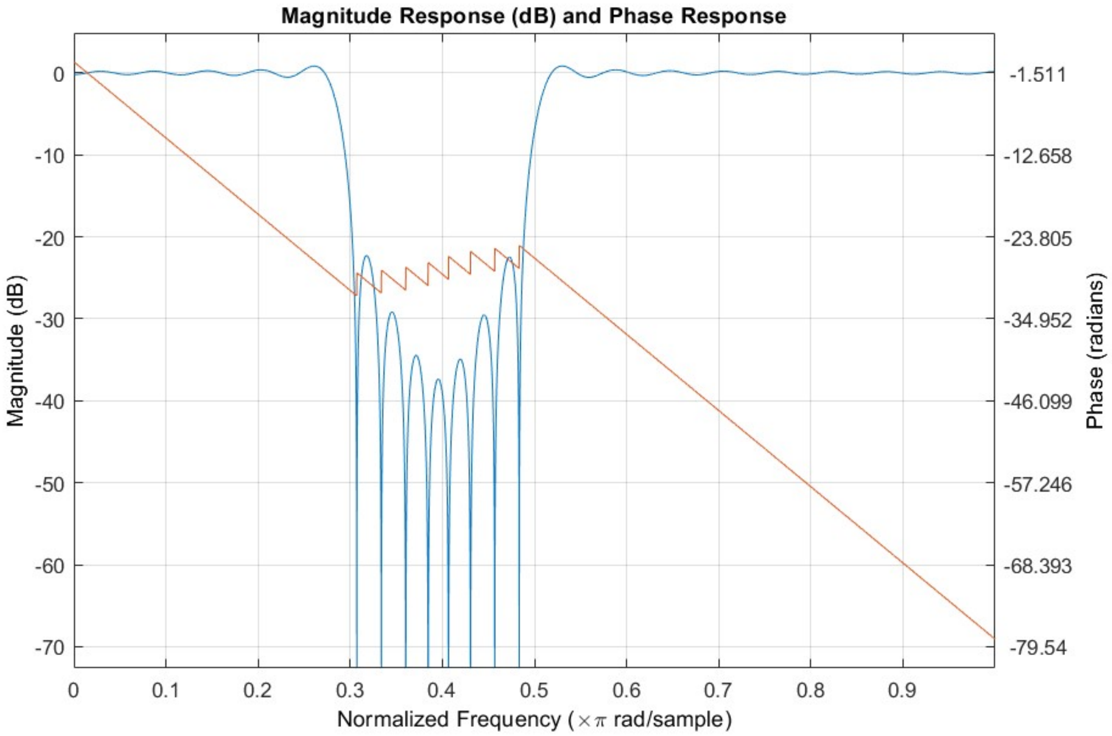
Here ω_T is the minimum transition width from the stopband to the passband.

$$\Delta\omega_T = \frac{5kHz \times 2 \times \pi}{425kHz} = 0.0232\pi \tag{6}$$

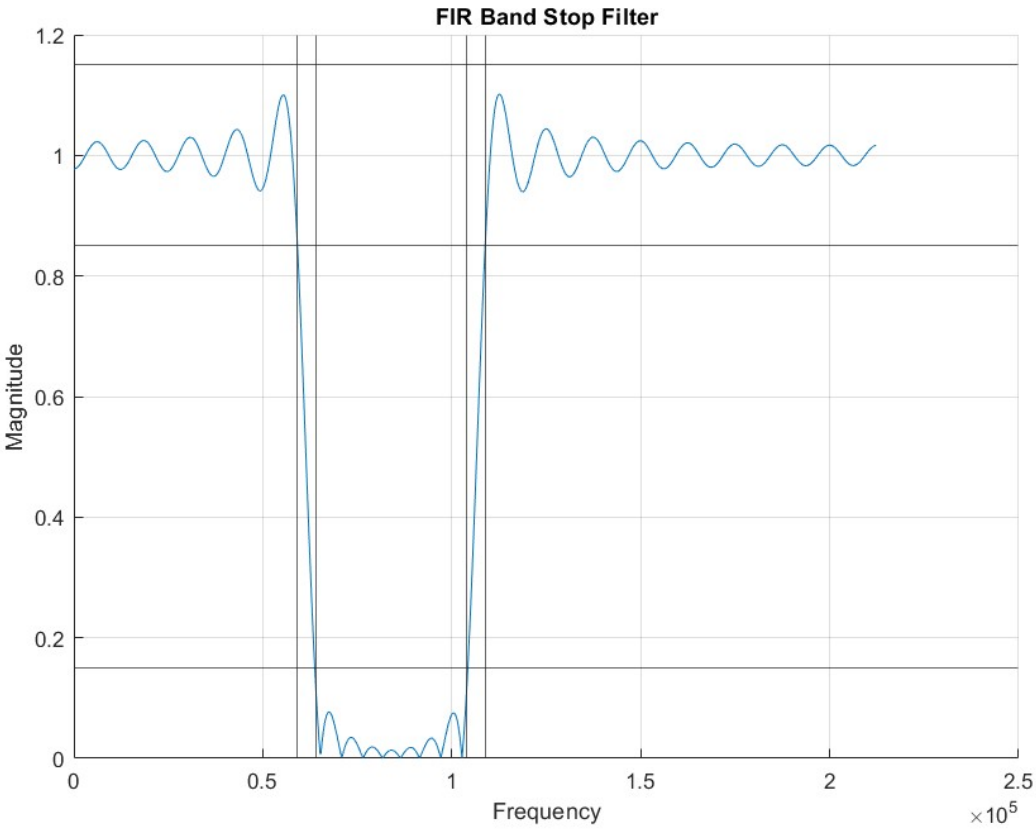
Substituting $\Delta\omega_T$ we get $N_{min} = 51$

The above equation gives a loose bound on the window length when the tolerance in not very stringent. From trial and error in MATLAB and checking if the specific given for the filter that I had to design the window length after this was **67**. The window is a rectangular window as $\beta = 0$.

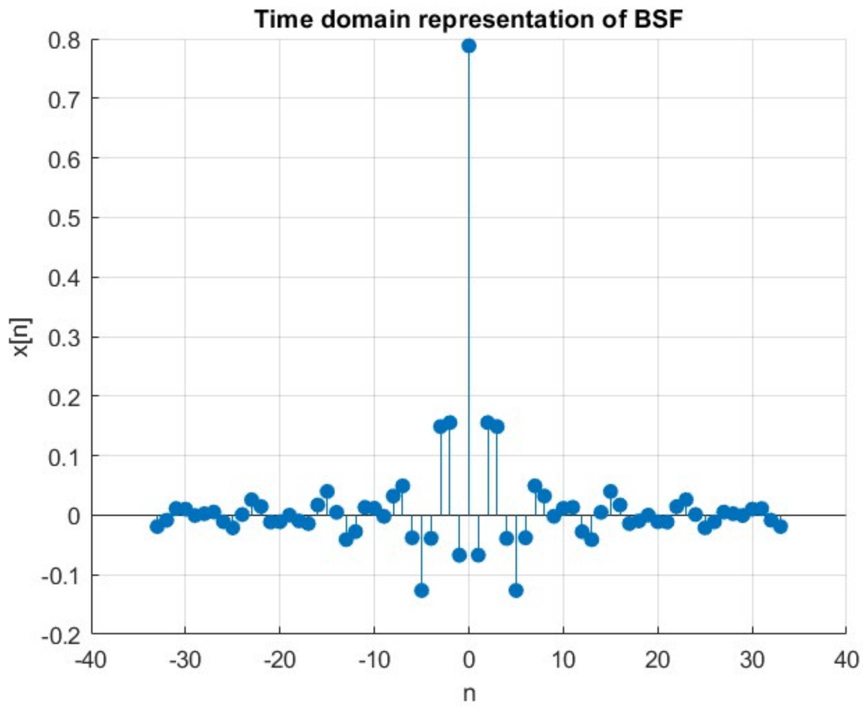
The time domain coefficients were obtained by first generating the ideal impulse response samples for the same length as that of the window. The Kaiser window was generated using the MATLAB function and applied on the ideal impulse response. The ideal impulse response was generated using a linear combination of impulse response of the ideal lowpass filter. The cutoff value and number of samples were given as the input argument to the ideal lowpass filter generating function.



Mynd 5: Magnitude Phase Response



Mynd 6: Frequnecy Response



Mynd 7: Impulse Response

```
FIR_BandStop =

  Columns 1 through 12

   -0.0191   -0.0085    0.0111    0.0101   -0.0006    0.0025    0.0052   -0.0110   -0.0213    0.0011    0.0260    0.0144

  Columns 13 through 24

   -0.0115   -0.0110   -0.0000   -0.0096   -0.0140    0.0171    0.0398    0.0049   -0.0411   -0.0276    0.0131    0.0116

  Columns 25 through 36

   -0.0019    0.0321    0.0492   -0.0379   -0.1264   -0.0390    0.1488    0.1555   -0.0672    0.7882   -0.0672    0.1555

  Columns 37 through 48

    0.1488   -0.0390   -0.1264   -0.0379    0.0492    0.0321   -0.0019    0.0116    0.0131   -0.0276   -0.0411    0.0049

  Columns 49 through 60

    0.0398    0.0171   -0.0140   -0.0096   -0.0000   -0.0110   -0.0115    0.0144    0.0260    0.0011   -0.0213   -0.0110

  Columns 61 through 67

    0.0052    0.0025   -0.0006    0.0101    0.0111   -0.0085   -0.0191
```

Mynd 8: Coefficients for discrete impulse response

2 Peer Review Acknowledgement

To whosoever it may concern, I have succesfully reviewed this filter design report done by Hemant Hajare as a part of this course assignment for EE338, Digital Signal Processing. This includes the design process, context, calculations, simulation program and results.

Thank you,
Regards
Tejaswee Sulekh
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