FIR and IIR Filters

1- Consider the FIR filter $H(z) = b(1) + b(2)z^{-1} + b(3)z^{-2} + \cdots + b(31)z^{-30}$. The coefficients of **b(i)** are provided in the **"FIR.mat"** file. Plot the amplitude and phase impulse response of the filter.

2- Filter the signal **x** of part 2 in CA#1 with the given FIR filter and call the output **xFIR**, then using your own **"reconst"** function,try to reconstruct the signal **yFIR** from **xFIR**. Plot **yFIR** and x_a in the same figure and compare them.

3- Compute the delay value of the given FIR filter. Now ignore (subtract) the delay of **yFIR** and then compare the delay free reconstructed signal with x_a to see the quality of reconstruction. (Plot the signals in the same figures, one for phase and one for amplitude).

4- Consider the following IIR filter,

$$\boldsymbol{H(z) = 0.0781 \frac{b_1(1) + b_1(2)z^{-1} + b_1(3)z^{-2}}{a_1(1) + a_1(2)z^{-1} + a_1(3)z^{-2}} \times \frac{b_2(1) + b_2(2)z^{-1} + b_2(3)z^{-2}}{a_2(1) + a_2(2)z^{-1} + a_2(3)z^{-2}} \times \frac{b_3(1) + b_3(2)z^{-1} + b_3(3)z^{-2}}{a_3(1) + a_3(2)z^{-1} + a_3(3)z^{-2}}}$$

The coefficients are given in the file **"IIR.mat".** Plot the amplitude and phase of the given IIR filter.

5- Filter the signal **x** of part 2 in CA#1 with the given IIR filter and call the output **xIIR** , then using your own **"reconst"** function, try to reconstruct the signal **yIIR** from **xIIR** . Plot (the amplitude and phase of) **yIIR** and x_a in the same figure and compare them.

6- Try to manually shift **yIIR** to best match $x_a(t)$**.**

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Filter Design

8- Read the sound file namely **“NoisySound”,** and determine the sampling frequency. Play the sound file. As evident, the file is corrupted with a single tone noise.

9- Using **“fft”** function, plot the magnitude of the imported file in the frequency interval of $[0, Fs/2]$ where Fs is the sampling frequency of imported signal.

10- After plotting the fft and quantifying the interfering frequency, design a filter to eliminate the interferer. For your filter implementation use a band-stop FIR filter with minimum order to filter the interference. Choosing the proper values of cutoff frequency and maximum losses is up to you; try to design a filter with as low as possible order. Plot the magnitude and phase of the filter.

11- Using sound function play the noise-free sound. Then, write it to a file named **“noiseless.wav”.**

12- Use noiseless.wav and pass it through a digital FIR low passfilter with the following specifications:

- Approximation: Equiripple
- Passband cutoff frequency: 2000 Hz
- Stopband cutoff frequency: 2500 Hz
- Order: 35

a. Play the sound before and after the filter.

b. Plot the waveforms in the time domain before and after the filter. Can you tell intuitively what has happened to the time domain signal after the filtering process?

c. Plot the spectrums (fft magnitude) before and after the filter.

d. Export the filter coefficients into your workspace and plot the magnitude and phase of the filter.