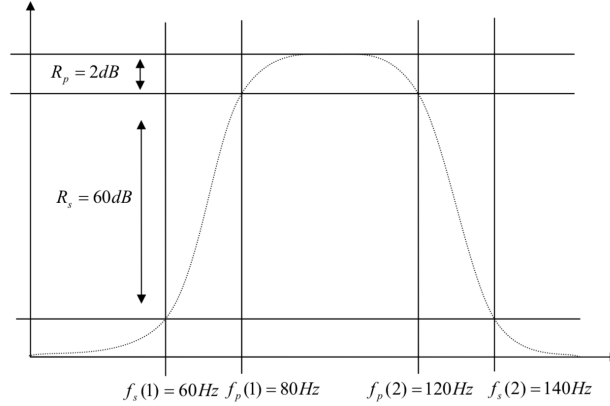


Lab 10

EE 324: Signals and Systems II

FIR and IIR digital filters. We aim at designing both FIR and IIR digital filters that satisfy the requirement in the figure below, namely, band-pass filter with central frequency $f_0 = 100Hz$ and bandwidth $B = 40Hz$.



1 Prelab assignment

1. Get familiar with the `fir1` function in matlab <https://www.mathworks.com/help/signal/ref/fir1.html>
2. Pick a proper sampling frequency f_s . Note that it has to be at least twice as fast as the signal frequency we are interested in, which has frequency between $f_0 - B/2 = 80Hz$ and $f_0 + B/2 = 120Hz$. A good candidate is $f_s = 300Hz$.
3. Calculate the normalized frequencies $\Omega_1, \Omega_2, \Omega_3, \Omega_4$ from $f_s(1), f_p(1), f_p(2), f_s(2)$ using $\Omega = 2\pi f/f_s$.
4. Start with an ideal band-pass filter

$$H_d(e^{j\Omega}) = \begin{cases} e^{-j\Omega \frac{M}{2}} & \Omega_2 \leq |\Omega| \leq \Omega_3 \\ 0 & \text{otherwise.} \end{cases} \quad (1)$$

Choose $M = 50$ and derive the impulse response h_d of H_d . One can then have many FIR filters based on the choices of window functions. Plot the impulse responses when the window function is rectangular.

2 Lab assignment

1. FIR
 - i). Design a FIR filter directly using `fir1` function in matlab with the same parameter $M = 50$. Compare its impulse response with the one you obtain in Prelab problem 4.
 - ii). Design FIR filters with different $M = 5, 10, 50, 100, 500$ and plot their frequency responses $H(e^{j\Omega})$ on the same figure. Describe your observations.

iii). Design FIR filters with $M = 50$ but with different window functions (rectangular, Hamming, hann). Plot their frequency responses on the same figure. Describe your observations.

2. IIR

i). Design a band-pass analog Chebyshev filter $H_a(s)$ with frequency characteristic as in the figure.

ii). Apply Tustin transform to obtain a digital filter.

iii). Using the function `freqz`, look at the frequency characteristics of the filter, and report the attenuation at the “frequencies” corresponding to $[60, 80, 120, 140]Hz$. Recall that in discrete-time Fourier transform (DTFT) domain, the “frequency” value 2π corresponds to the sampling frequency $f_s = 1/T_s$ in Fourier transform (FT) domain.

iv). You will note that the discretized filter fails to meet the required specifications. This is due to the frequency warping introduced by the Tustin transformation. Therefore, we need to design an analog filter based on the frequencies after prewarping

$$\omega = \frac{2}{T_s} \tan \frac{\Omega}{2}.$$

v). Apply Tustin transform to the analog filter established in iv) to obtain a digital filter. Verify the specifications of this new digital filter.

vi). Design a digital filter directly using the Matlab commands `cheb1ord` and `cheby1` (but without mentioning the argument ‘s’). Note that the frequencies used as the arguments of `cheb1ord` should be normalized to the Nyquist frequency. Visualize its frequency characteristics and compare them to the filter you designed by “manual” prewarping in v).

3. Simulation

Apply your digital filters to the discrete-time version of the signal $x(t) = \sin(200\pi t) + \sin(120\pi t) + \sin(280\pi t)$. Compare their performances and describe your observations.