EE432 Fall 2011 Project 08: Approximating an Ideal LPF (Due: 11/03/2011)

In this project, you will explore create an FIR filter that approximates an ideal low pass filter, then evaluate some of the characteristics of its frequency response.

I. FIR Approximation to an Ideal LPF Filter Design

1. ~~Write a MATLAB function that implements a digital sinc, called *dsinc.m*. The usage is:~~

   Step 1 for Fall 2012

   ~~*y=dsinc(n,omega)*~~

   ~~Where *y* is the output sinc values, *n* is an input vector over which the digital sinc is to be computed, and *omega* is the digital sinc's cutoff frequency. For error checking, use the *round* function to ensure that input *n* contains all integer values.~~

2. Create a 51-term digital sinc-shaped impulse response (call it $h_1[n]$) for an ideal low pass filter with cutoff frequency $\Omega_1 = 0.35\pi$. The 51 terms should be centered around n=0. On a 3x1 subplot, use *stem* to plot this on the upper plot with properly labeled axes.

3. Truncate this sinc so that it contains only 15 non-zero terms (call this $h_2[n]$) …this means that there should be 51 terms, but there are only **7** non-zero terms to the left and right of n=0. On the same 3x1 subplot, plot this impulse response on the middle plot, again with proper labels.

4. Now time shift $h_2[n]$ so that it is causal (call this impulse response h[n]). Plot h[n] on the bottom plot in the 3x1 subplot, and turn this plot in. Note: this impulse response should start at n=0. You'll use this impulse response in the analysis that follows.

II. Analysis of Frequency Response

1. By whatever means you choose, create a plot of the magnitude response and the phase response of the filter. Determine the following values for this filter by filling in the table that follows, assuming a sampling frequency of **44.1 kHz**. Indicate on your plot how you determined your answers. Note, $G_{dB} = 20 \log_{10}(G)$.

   For the frequency response, you can either:

   a. Determine the difference equation and then input the b_n and a_n coefficients into *fvtool* and display the magnitude plot to get your answers Note: the plot is in dB, so you'll have to undo decibels to fill in part of this table (can change from dB under "Analysis Parameters").

   OR

   b. Write a function to compute the DTFT for values of radian frequency for $0 \le \Omega \le \pi$. Plot the magnitude and use this plot along with sample frequency to determine the answers.

# Sidelobes in $H(\Omega)$		
Cutoff Frequency (Hz)		
Filter bandwidth (Hz)		
Pass band edge frequency (Hz)		
Stop band edge frequency (Hz)		
Transition width (Hz)		
Maximum Gain, $1+\delta_p$	V	dB
Pass band ripple δ_p	V	dB
Stop band ripple δ_s	V	dB

2. Now repeat the steps above using a 41-term filter…that is, the 51-term sinc will be truncated and have 41 non-zero terms, so the filter will also have 41 non-zero coefficients. Fill in the table below:

# Sidelobes in $H(\Omega)$		
Cutoff Frequency (Hz)		
Filter bandwidth (Hz)		
Pass band edge frequency (Hz)		
Stop band edge frequency (Hz)		
Transition width (Hz)		
Maximum Gain, $1+\delta_p$	V	dB
Pass band ripple δ_p	V	dB
Stop band ripple δ_s	V	dB

3. Describe what effect that the number of non-zero terms has on the filter's characteristics.

For this project write up, turn in your code, the plots of impulse response, the plots of magnitude response annotated with indications of the values you entered in the tables above (hand drawn is fine), the tables above, and answer the last question.