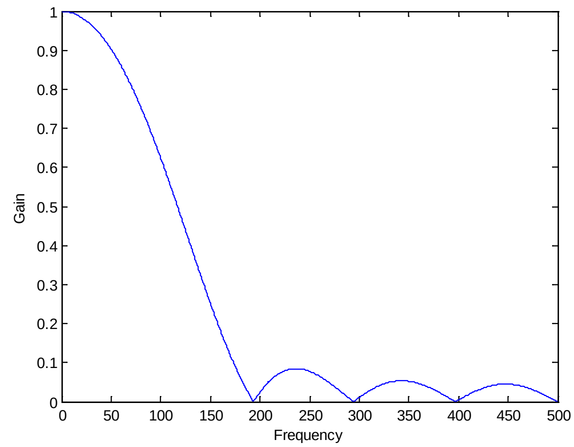
EE 311 **February 17, 2017**
FIR Window problems

14. The figure below is that of a low pass filter from Matlab.

A) What is the minimum number of zeros that are on the unit circle for this filter? Explain your answer.



B) What is the minimum order of this filter. Explain your answer

16. Suppose you have a low pass filter with a cut off frequency of f_c and a sample frequency of f_s. If you double the sample frequency and keep the cut off frequency in the same place do the poles and zeros get closer together, further apart, or neither. Justify your answer.

5.5 A triangular window function of length 5 has an impulse response given by

$w_{Tri}(n) = \{0.333, 0.667, 1.0, 0.667, 0.333, 0, 0, \cdots\}$

An ideal low pass filter has an impulse response given by:

$h_I(n)$ = {…, 0.2, 0.3, 0.6, 0.8, **1.0**, 0.8, 0.6, 0.3, 0.2, …}

where the bold font indicates the n = 0 term. Find the transfer function for the windowed filter.

5.9 The impulse response for an ideal filter is used to create a windowed digital filter. The impulse response is given by:

$$h(n) = \sum_{n=-\infty}^{\infty} \frac{1}{2^{|n|}}$$

The final filter designed with this impulse response has a transfer function given by:

$$H(z) = \frac{z^4 - 3z^3 + 2z^2 - 3z + 1}{z^4}$$

Find the window function, W(n) used to create this filter.

5.13 Begin with an ideal lowpass filter which has a cutoff frequency of 3,000Hz and create 5 digital FIR filters with a sample rate of 20,000Hz. These five filters should be of order 30 and use different window techniques including rectangular, Bartlett, Hamming, Von Hann, and Blackman. For each filter determine the transition band width which is defined as that band of frequencies whose gain is between .1 (10%) and .9 (90%). Compare this transition band width to the main lobe width data of Figure 5.17.

5.22 In an introductory lecture on FIR filters a professor asked the class to consider how a band pass filter could be produced. A fuzzy student in the back row who had slept through most of the chapter on frequency analysis and z-transforms responded that a band pass filter could be produced by sampling a sine wave whose frequency corresponded to the desired center frequency of the filter and using the sampled values as the filter coefficients. Thus, if the center frequency is 1KHz and the sample frequency is 12KHz, we would produce the numerator coefficients b_0 to b_{12} by sampling a sinusoid as shown in the figure P5.12 When this sinusoid comes through the filter, it's value will therefore be squared and all other sinusoids will have some lesser value. Is this a valid argument? Use MATLAB® to produce an FIR filter to verify your claim.

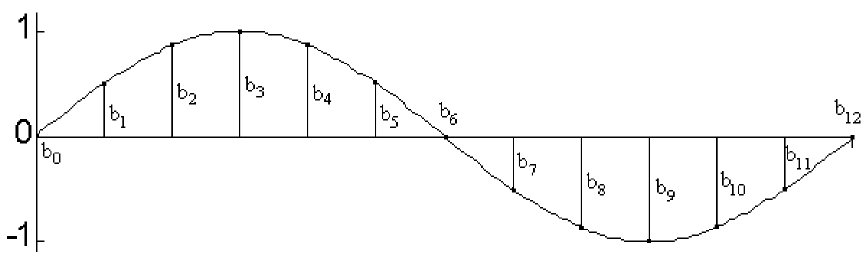


Figure P5.22
Does a sampled sinusoid lead to a band pass filter?