EE 311
Final Exam Review 2
Spring 2013/14

Final Exam is Friday, May 2, 2014 at 8:00am

1. Explain how you can use MATLAB® to quantize the coefficients of a filter.

2. Suppose that an analog temperature sensor produces output voltages to indicate temperature in the range of -30°C to +100°C with an accuracy of ±0.15°C. How many bits are needed in the A to D converter to make use of the full range and accuracy of this sensor? (Show your work.)

3. Suppose that an analog temperature sensor produces output voltages to indicate temperature in the range of -30°C to +100°C. If a 10-bit A to D converter is used to read the signal what will be the error range for the signal read in (in degrees). (Show your work.)

4. Suppose I have a filter section such that if I put in an input with an amplitude of 1 at some frequency f_x the output has an amplitude of 0.7 and is shifted in phase by -40°.

A) If I put two such sections in *series*, what will the gain and phase be for a signal at frequency f_x? (Show your work.)

B) If I put two such sections in *parallel*, what will the gain and phase be for a signal at frequency f_x? (Show your work.)

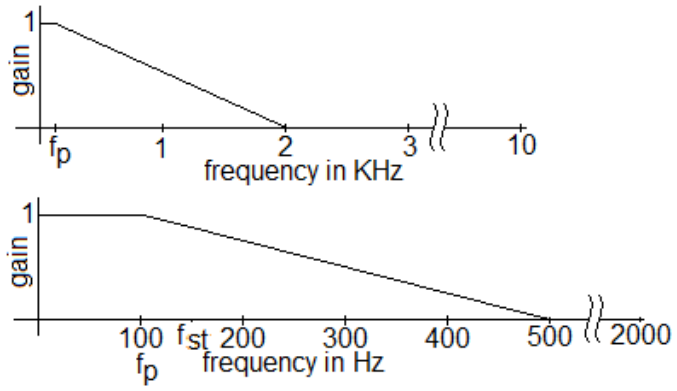
5. When a signal is input to a digital filter it is quantized to b-bits by the A to D converter. How does this affect the phase shift of the filter. Justify your answer.

6. Quantization error can be reduced simply by increasing the number of bits used to represent coefficients and variables. Give two reasons why is this not an option in many practical systems.

7. When we up-sample a signal we do so by first inserting zeros between the existing samples and follow with a low pass filter which "fills in" the zeroed sample points. Are we creating new information? If so, does it accurately reflect the true value of the signal. Is there any difference between say, sampling a signal which is band limited to 0 to 100 Hz at 1 KHz and sampling the same signal at 500 Hz and using an interpolator to up-sample it to 1 KHz?

8. Show that for a cascade of N decimation filters each of which has a pass band ripple of R_{p1}, the decimated signal will have a ripple given by $R_p \approx NR_{p1}$.

9. In Example 7.2, we did down-sampling by a factor of 4 and 5 and we used two low pass filters whose pass and stop bands are shown in Figure below. If the pass and stop band ripple for these two filters are designed to be 0.01, calculate the ripple to be expected for the pass and stop and of the final cascaded filter.



10. Suppose you have two identical signals which are limited to a small band of frequencies around 1000 Hz. The first signal is sampled at 11025 Hz and the second is sampled at 22050 Hz. If we up-sample both signals to 44100 Hz what will be the difference in the frequency characteristics of the two.

11. A signal which is limited in frequency to the band from 0 to 4 KHz is sampled at 11,025 Hz. If a 8-bit A/D converter is being used and we want to up-sample this signal to 22,050 Hz, what are reasonable specifications for the interpolation filter if we want to use a Parks-McClellan filter.