EE 311 **February 1, 2016**

Exam 1 Review

1. Explain why one would expect an IIR filter to be computationally more efficient for a given algorithm than an FIR filter.

2. FIR filters do not have a feedback path. What are the implications of this for system stability? Is oscillation possible? Why or why not?

3. IIR filters have a feedback path and FIR filters do not. What would you expect to be the result of small errors in the formulation of the coefficients for these two systems? Would the impact of such an error be greater or less for an IIR filter? Explain.

4. Trapezoidal integration was used to approximate an integral and convert a differential equation to a difference equation. What would be the consequences of using an approximation based on a second order equation such as a parabola rather than the straight line fit of the trapezoidal method?

5. Suppose that in the evaluation of a difference equation the coefficients of the input variable and the delayed versions of the input variable are multiplied by a constant. What is the consequence of this multiplication in the frequency domain?

6. Suppose the impulse response function for an FIR difference equation is symmetric. For example the response might be given by
$h(nT) = \{b_0, b_1, b_2, b_3, b_4, b_5, b_4, b_3, b_2, b_1, b_0\}$.
How can the difference equation be written to use this symmetry to reduce the number of multiplications necessary for its evaluation?

7. When the coefficients of a difference equation are implemented in real time we would like to represent the coefficients with as many bits as possible since this reduces the quantization error. Give three reasons why increasing the number of bits has negative consequences on the implementation of a difference equation.

8. In general a sinusoid can be represented by the equation $y = A\sin(2\pi ft)$ where A is the amplitude and f is the frequency. This equation has two unknowns so that at least two values for y at two different time samples are needed in order to determine A and f. What other information do I need in order to *uniquely* determine values for A and f?

9. If a continuous-time signal is sampled and the sample values applied in a digital computer, why is the sample value used in the computer generally not equal to the value of the continuous-time signal at the sampling instant.

10. Discuss the difference between the continuous-time impulse $\delta(t - t_0)$ and the discrete-time impulse $\delta[n - n_0]$.

11. Describe, in English, the process of convolution with respect to a signal being processed through a system.

12. Why is it important to use orthogonal basis functions for signal representations such as the Fourier series?

13. Considering that $\sin(0) = 0$, why does $\mathrm{sinc}(0) = 1$?

14. What is the relationship between the Fourier series and the Fourier transform?

15. What is the relationship between the Fourier transform and the Laplace transform?

16. What is the relationship between the Laplace transform and the z transform?

17. What is the relationship between the DTFT and the z transform?

18. The Fibonacci sequence is given by {0 1 1 2 3 5 8 13 21 33 54 ...} where each term after the first two is the sum of the previous 2 terms.

A) Find the difference equation for the sequence.

B) Write the z transform from the difference equation.

19. Suppose that you have designed a digital filter to have a cutoff frequency of 200 Hz and a sample frequency of 1000 Hz. In implementing the filter, an error is made and the sample frequency becomes 2000 Hz. What happens to the cutoff frequency. Justify your answer.

20. Suppose a system has a sample frequency of 1000 Hz. If you input a sinusoid of frequency of 1300 Hz what will be the main frequency of the output?

21. Suppose I have a band pass filter and I apply a step input. Since after t = 0 the input will always be 1, will the output ever go to zero? Explain.

22. In the trigonometric form of the Fourier series the dc term shows up as $a_0/2$. How is the dc term expressed in the exponential form of the series?

23. Suppose you construct what is supposed to be a high pass filter in the lab. After getting things together you apply a unit step to the filter and notice that the output, after an initial transient, settles down to 2 volts. Can you conclude that your filter is working correctly or not. Explain.