CMPE323 Fall 2014 Final Exam   Student ID # ________________________________
12/5/2016

Instructions:

1) **This is a take-home exam, and it is critical that you follow these directions exactly.**
2) **The UMBC policy on academic integrity will be strictly enforced. I will disqualify your exam on the slightest hint that the rules have been violated.**
3) **Make sure your student ID # is on the cover sheet and be sure your submission is securely stapled in the upper right corner.**
4) **You may use any notes from this course, any homework solutions, and the textbook. You may use external references *if they are accompanied by a properly formatted citation and reference* as indicated in the examples posted on Blackboard.**
5) **You may *not* consult with any human other than Dr. LaBerge, who will provide clarifications to questions submitted on the Blackboard Discussion thread until 6 PM on Sunday December 18. All requests for clarifications will be posted to the entire class, so all are on a common basis. You may consult with pets that live in your domicile.**
6) **You may use MATLAB in any manner you desire. Please submit any and all of your MATLAB files (scripts or special-purpose functions) so that I can be assured that a) you did the work and b) the MATLAB you turn in gives me the exact plots and results that you turned in. MATLAB submission must be on Blackboard. I will be comparing MATLAB files, so do your own work. I will have a Blackboard submission site for the MATLAB files, do not submit hard copies of the MATLB scripts or special purpose functions. Do submit MATLAB plots and relevant print outs in hard copy.**
7) **The exam (other than MATLAB files) must be submitted in hard copy in my office, ITE358, by 7 AM on Tuesday, December 20. *Please be sure your work is legible* and the steps you take in your solution are well documented so I can follow your logic.**
8) **Although I'm usually lenient, *I will not accept late exams* in this case. You can't afford for this exam to not count.**

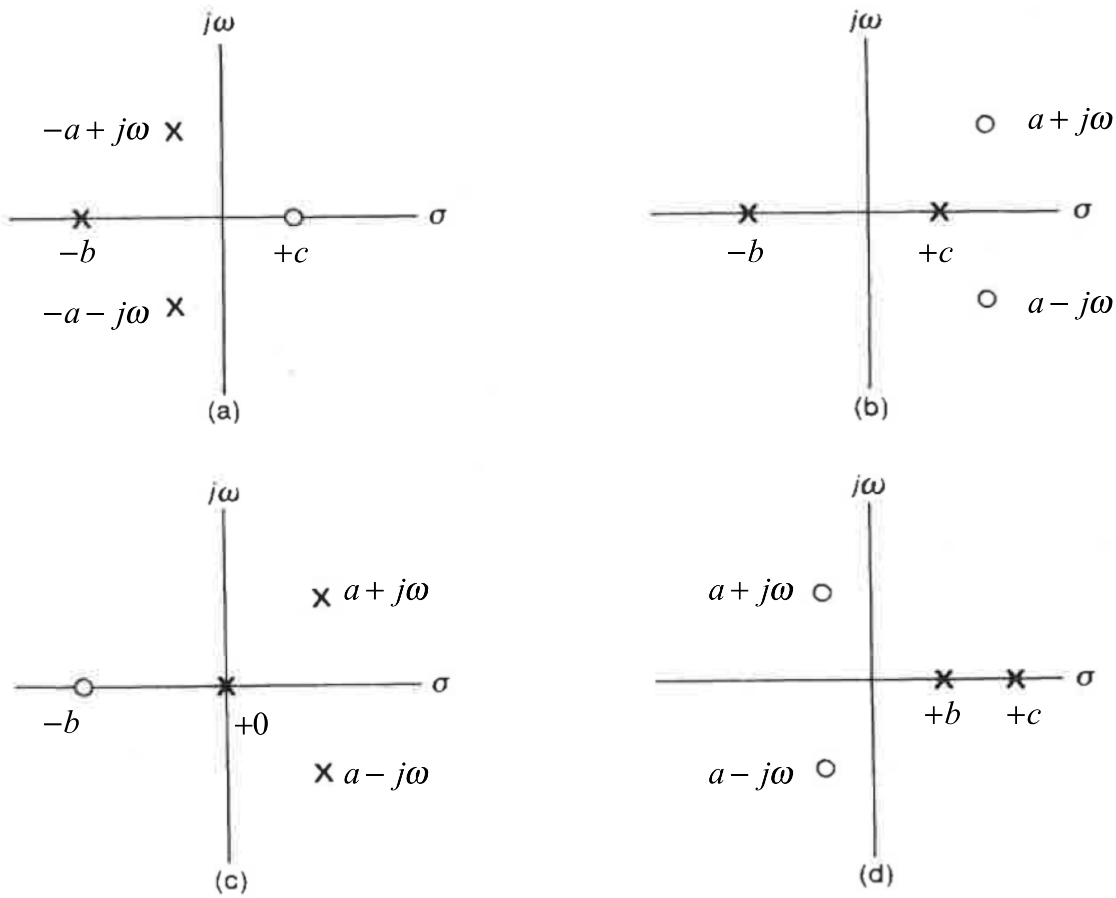
By submitting this exam, including the MATLAB portions, you agree to be bound by the constraints of the UMBC Policy on Academic Integrity and you confirm that you *have read, understood, and complied with the rules listed above*. Your submission indicates that you agree with this condition. Penalties for violations will be severe.

There are 110 points on Part I and 100 points on Part 2 of this exam. You can only earn 100 points in each section.

GOOD LUCK

PART 1: Analytical Work (110 points)

1. (20 points) For each of the following pole/zero plots show the RoC for stability. Based on the RoC and the information given, write the impulse response for each system. You may assume that $a,b,c,\omega > 0$

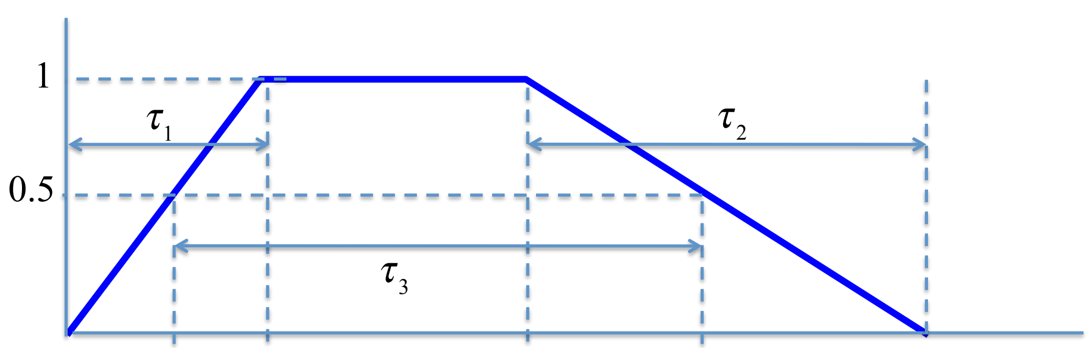


2. (10 points) Consider a system that is a simple integrator, $y(t) = \int_{-\infty}^{t} x(\tau)d\tau$. Prove that this system is linear, time-invariant, causal, and non-static. Determine if the system is BIBO stable; if it is not, find a bounded input that creates an unbounded output. Find the unit step response and from it the impulse response of this system.

3. (10 points) Define a *unit area pulse* $a(t;\tau) = \begin{cases} 0 & t < -\frac{\tau}{2} \\ \frac{1}{\tau} & -\frac{\tau}{2} \le t < \frac{\tau}{2} \\ 0 & t \ge \frac{\tau}{2} \end{cases}$

   Define two different inputs to the simple integrator to be $x_1(t) = a\left(t + \frac{\tau_3}{2};\tau_1\right)$ and $x_2(t) = a\left(t - \frac{\tau_3}{2};\tau_2\right)$, where τ_3 is a positive constant that will be used later. Use convolution to determine analytically the associated outputs $y_1(t)$ and $y_2(t)$ to the inputs $x_1(t)$ and $x_2(t)$, respectively. Plot the two responses (you may use MATLAB).

4. (10 points) Consider the asymmetrical trapezoidal waveform, $w(t;\tau_1,\tau_2,\tau_3)$ shown below. The waveform has unity amplitude, and is defined by three straight-line segments: the rise time (τ_1), the half-voltage pulse width (τ_3), and the fall time (τ_3). Show that $w(t+\beta) = y_1(t) - y_2(t)$, where $\beta = \frac{(\tau_1 + \tau_3)}{2}$



5. (20 points) Find the Fourier transform $W(f)=\mathcal{F}\left(w(t)\right)$ using the convolution theorem and the linearity of the Fourier Transform. For full credit, your answer must be expressed in terms of exponentials, powers of f and sinc functions, where $\operatorname{sinc}(x)\triangleq\dfrac{\sin(\pi x)}{\pi x}$ (as implemented in MATLAB)

Hint: You may use the fact that, for all practical purposes, $r(t)\delta(t-a)=r(a)$ for a function that is continuous at a.

6. (20 points) Find the Fourier transform $W(f)=\mathcal{F}(w(t))$ (same as last problem), using differentiation and the integration property of Fourier Transforms. If your answer differs from that obtained in Part 5, explain why.

7. (10 points) Using your answer to either 5 or 6, verify your answer by showing that it converges to the expected result in the case where $\tau_1=\tau_2,\ \text{and then}\ \tau_1\to 0$. For full credit, you must explain why your "expected result" is, in fact, expected.

8. (10 points) Using your answer to either 5 or 6, verify your answer by showing that it converges to the expected result in the case where $\tau_1=\tau_2=\tau_3$. For full credit, you must explain why your "expected result" is, in fact, expected.

Part 2: Simulation Work (100 points)

We're going to revisit the Double Sideband Amplitude Modulated waveform from Lab 1. As a reminder, our DSB-AM waveform has the form

$$d(t)=A\left(1+a_1\cos(2\pi f_1 t+\phi_1)+a_2\cos(2\pi f_2 t+\phi_2)\right) \tag{1.1}$$

(in lab we had three terms, but two is enough for this problem). In (1.1), a_k is an amplitude factor, f_k is the frequency of modulation, ϕ_k is the relative phase of the modulating component at $t=0$, A is an overall amplitude factor, and, for this example, $k=1,2$ for the two modulating terms. DSB-AM is a means of communicating an *analog* waveform. In this case, the analog waveform is the sum of two sinusoids.

9. (10 points) Let $A=1,\ a_1=a_2=0.2, f_1=90\ \text{Hz}, f_2=150\ \text{Hz},\ \phi_1=\phi_2=-\dfrac{\pi}{2}$. Choose and state a simulation rate appropriate for simulating $d(t)$ as an analog waveform. Using that rate, compute and plot $d(t)$ for at least three cycles of the combined periodic waveform. Use professional practices in your plots.

10. (15 points) Choose a sample rate (possibly different from part 9) and appropriate sample size that permits you to perform an accurate FFT of your waveform $d(t)$ with resolution of exactly 10 Hz. Using

this sample rate and number of samples, estimate the voltage spectrum (Fourier Transform) $D(f)$ of the waveform.

11. (15 points) Create a new signal, $x(t) = d(t)\cos(2\pi f_C t)$, where the DSB-AM waveform is used to amplitude modulate a carrier signal with $f_C = 100\text{ kHz}$. Choose a sample rate and appropriate sample size to perform an FFT of your modulated waveform $x(t)$ with a resolution of exactly 10 Hz. Plot the modulated waveform, and the magnitude of the voltage spectrum $X(f)$ in the ranges 98-102 kHz and minus 102 to minus 98 kHz. Which Fourier Transform property is illustrated by the fact that there are there versions of the baseband spectrum, $D(f)$, around $\pm 100\text{ kHz}$? Compare the phases of $X(f)$ in the same band and explain why they have the values you see using the properties of the Fourier Transform. Using this amplitude spectrum in the vicinity of 100 kHz, explain why this waveform is called "double sideband AM".

12. (30 points) Return now to the "baseband" version of the waveform, $d(t)$. (Without the carrier). This problem considers the case where both the desired signal, $d(t)$ and an undesired signal on an adjacent channel are present at the input to the signal processing. For the purpose of this problem, an adjacent channel signal is a modulated signal (like 11) with $f_C =$ 50 kHz. The adjacent channel signal has the parameters $a_{1\text{A}} = 0, a_{2\text{A}} = 0.4, f_{1\text{A}} = 90\text{ Hz},\ f_{2\text{A}} = 150\text{ Hz}, \phi_{1\text{A}} = \phi_{2\text{A}} = 0$, where the subscript "A" stands for "adjacent channel". For the purposes of this problem, assume that the adjacent channel signal is signal input is 34 dB *higher* than the desired signal. With these assumptions, define an appropriate filter to reduce the first adjacent channel signal to 30 dB *below* the desired signal (you can assume $A = 1$ for the desired signal.) Provide the class of filter (Butterworth, Chebyshev Type I, Chebyshev Type II), and the number of poles in the lowpass equivalent filter. **For this problem, you need only design the low pass filter!!** You must assume that the center frequency of the desired signal may vary over the range of $0\text{ to }10\text{ kHz}$. [1](This helps determine the filter bandwidth.) *Hint: Signal amplitudes are given in volts, not watts.*

13. (15 points) Provide a pole-zero plot of the filter transfer function $H(s)$ of your low-pass equivalent filter. Use MATLAB to plot the frequency response of the filter transfer function $H(s)$. Your plot must address the magnitude of the filter transfer function in decibels. (Consider the MATLAB function `freqs`, but do your own scaling. A scale of hundreds of decibels is not acceptable.) Comment on the number of poles and their location and on the shape of the magnitude of the transfer function.

14. (15 points) Perform a Partial Fraction Expansion of $H(s)$ to determine the analytical form of the filter impulse response, $h(t)$ of the *low pass equivalent* filter.

[1] By "vary over", I mean that the baseband signal is really $d(t)\cos(2\pi f_B t)$, where $0 \le f_B \le 10\text{kHz}$. For the $d(t)$ given in Problem 9, $f_B = 0$. The value of f_B represents the possible frequency error of the transmitter.