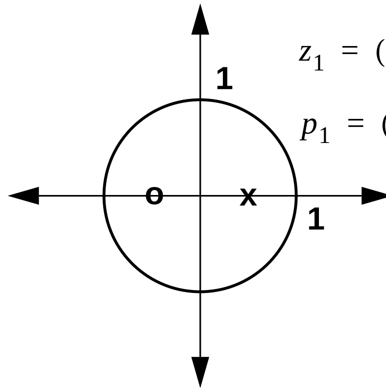
Name:

Problem	Points	Score
1	10	
2	10	
3	10	
4	10	
5	10	
6	10	
7	10	
8	10	
9	10	
10	10	
Total	100	
Extra Credit	15	

Notes:

1. The exam is open books/open notes.
2. Please show ALL work. Incorrect answers with no supporting explanations or work will be given no partial credit.
3. If I can't read or follow your solution, it is wrong, and no partial credit will be given — BE NEAT!
4. Please indicate clearly your answer to the problem.
5. Several problems on this exam are fairly open-ended. Since the evaluation of your answers is obviously a subjective process, we will use a marketplace strategy in determining the grade. Papers will be rank-ordered in terms of the quality of the solutions, and grades distributed accordingly.

1. (Is this the convolution problem?) A pole/zero plot for a linear system is shown to the right. Assume the gain of the system is 1.0. Construct a detailed plot of the impulse response, indicating the time at which the impulse response permanently stays below its value at $n = 0$. Relate this to the location of the pole in the z-plane.



$z_1 = (-1/2, 0)$

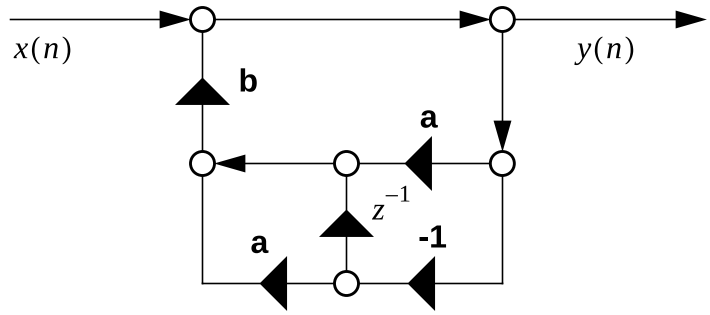
$p_1 = (1/2, 0)$

2. (Is this the convolution problem?) The system shown below is time-invariant. The input shown below is applied to the system, and the output is measured as shown. Determine the system impulse response, and whether the system is linear. You MUST justify your answers or no points will be awarded.

$$x(n) = \{1, 2\} \longrightarrow \boxed{H[\ ]} \longrightarrow y(n) = \{0, 2, 3\}$$

3. The signal flow graph below is noncomputable because it contains a closed loop with no delay elements. Obtain a flow graph for the system below that is computable (meaning that it has the same transfer function, but a different realization).

Note that the unfilled circles in the graph to the right denote adders (summers).



4. Design a system that uses a DFT and a rectangular window to detect one of the two signals shown to the right by examining the spectrum of the received signal (as described in class). Assume a sample frequency of $10\ kHz$. It is important that you minimize the overall complexity of the system.

$$x_1(t) = A\sin(2\pi \cdot 1000t)$$
$$x_2(t) = A\sin(2\pi \cdot 1007t)$$

5. Design the lowest order high-pass filter that has a cutoff frequency of 1 kHz, a sample frequency of 8 kHz, a passband gain of 1, a passband ripple of 1 dB, a stopband attenuation of 40 dB (with as much ripple as you want as long as the minimum attenuation is 40 dB), and a transition band of 500 Hz.

6. (Is this the convolution problem?) Explain how to compute the output for the system shown below using a DFT and a rectangular window. Explain any differences between this result and what you would get using convolution.

$$x(n) = \{1, 2\} \longrightarrow \boxed{H[\ ]} \longrightarrow y(n) = ???$$

$$h(n) = \{1, 2, 1\}$$

7. Show that if $c_{xx}(n) = \sum_{k=-\infty}^{\infty} x(n)x(n+k)$, then, $C_{xx}(z) = X(z)X(z^{-1})$.

8. (Is this the convolution problem?) Downsample the following signal from 8 kHz to 4 kHz using the simple low-pass filter shown for all filter designs:

$$x(n) = \{1, 1, 2, 2\} \qquad h(n) = \{1, 1\}$$

Be sure to explain each step.

9. (Perhaps this is the convolution problem???) You are given a system:

$$y(n) = x(n) + nx(n-1) \quad .$$

When you apply an impulse function to this system, you obtain the following output:

$$g(n) = \{1, 2, 0, 0\} \quad .$$

You are now given a black box. You apply the same impulse and measure the output to be the same as above. You now build a system by cascading the first system with the black box. You apply the sequence:

$$x(n) = \{1, 0, -1, 0, 0, 0, \ldots\} \quad .$$

Compute the output.

10. (This is not the convolution problem... guess you have to go back and look again.) List all the techniques we described in this course to determine the stability of a linear system.

(Extra Credit) This problem was included just to keep Julie Ngan happy. You will only be awarded points on it if your exam score on the previous 10 problems exceeds 60 points.

Describe one concept that you believe is truly unique to DSP (doesn't have a direct counterpart in analog system theory). Present the theory, discuss its impact, give a real world example of its use, and explain why this is unique to DSP.

(Optional) What grade do you think you deserve in this course? Explain. Point to your answers to some of the problems on this final as evidence you deserve such a grade.