New Mexico State University
Klipsch School of Electrical Engineering

EE395 - Introduction to Digital Signal Processing

Fall 2012
Exam #2 Part 1

Name: ____________________

Prob. 1		/ 17 points
Prob. 2		/ 17 points
Prob. 3		/ 16 points
Total		/ 50 points

Prob. 1

Compute the z-transform of the following signals. Include the Region of Convergence (ROC).

(a) $x(n) = 3\delta(n) + \delta(n-2) + \delta(n+2)$

(b) $x(n) = \left(\frac{1}{2}\right)^n u(n+2) + (3)^n u(-n-1)$

Prob. 1 (cont.)

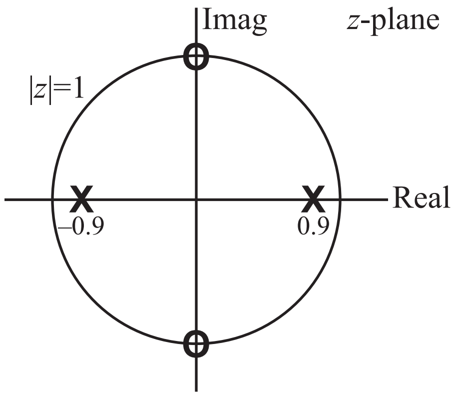
Compute the inverse of the following z-transforms.

(c) $X(z) = 4 + 3\,(z^2 + z^{-2}),\ \ 0 < |z| < \infty$

(d) $X(z) = \dfrac{1 + z^{-1}}{\left(1 - \dfrac{1}{2}z^{-1}\right)\left(1 - \dfrac{1}{3}z^{-1}\right)},\ \ |z| > 1/2$

Prob. 2

(a) A causal filter is described with the following pole/zero pattern. Determine the transfer function $H(z)$ and I/O difference equation.



(b) The causal filter is described with the following impulse response. Determine the transfer function $H(z)$, I/O difference equation, and pole/zero pattern.

$h(n) = \delta(n) - 0.0625\delta(n-4)$

Prob. 2 (cont.)

(c) A causal filter is described with the following I/O difference equation. Determine the transfer function $H(z)$, closed-form impulse response, and pole/zero pattern.

$$y(n) = 0.9y(n-1) + x(n) + x(n-1)$$

Prob. 3

(a) Draw direct form I and canonical (direct form II) realizations of the causal filter

$$H(z) = \frac{4 + \frac{9}{4}z^{-1} - \frac{1}{4}z^{-2}}{1 + \frac{1}{4}z^{-1} - \frac{1}{8}z^{-2}}$$

(b) Draw the cascade realization form of the causal filter

$$H(z) = \frac{1 + 2z^{-1} + z^{-2}}{1 - z^{-1} + \frac{7}{8}z^{-2}} \cdot \frac{1 + 2z^{-1} + z^{-2}}{1 + 2z^{-1} + \frac{3}{4}z^{-2}}$$

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Exam #2 Part 2

(Solution due in GA160G 10:30am Fri., Nov. 2, 2012)

"The attached solution is due entirely to my own, individual efforts. I have not communicated with any other student about this project nor have I consulted with anyone other than (possibly) the instructor of this course in creating these solutions."

Signature: ______________________________ Date: ____________________

Printed name: ____________________________

Prob. 1	/ 17 points
Prob. 2	/ 17 points
Prob. 3	/ 16 points
Total	/ 50 points

Directions

Please download the CompanionFilesExam2Part2.zip file

http://www.ece.nmsu.edu/~pdeleon/Teaching/EE395/Exams/CompanionFilesExam2Part2.zip

For each problem, please write a MATLAB code which produces the required plots. Name your codes main1.m, main2.m, and main3.m.

Items to be Submitted

Please submit the *signed* cover sheet of this exam and a printout of your solutions including plots, comments, and main codes. In addition, please send (before the due date) an email to `pdeleon@nmsu.edu` with a .zip file archive of codes *and* DSP tools. The archive should be organized with a folder (labeled with your last name) containing the three main codes and a subdirectory called "DSP_toolkit" containing your tools. You will receive a confirmation email upon receipt of the .zip file.

Due Date

Your printed solutions (plots, comments, and main codes) and electronic submission of your codes are due on or before **10:30am Friday, Nov. 2, 2012**.

Office Hours and Appointments

Extended office hours will be held on Thu., Nov. 1, 9:00–11:00am and 3:00–5:00pm. Students are encouraged to discuss the problems with Prof. De Leon. In addition, you are free to email any questions to Prof. De Leon during the project period. Email may include a request to examine code.

Prob. 1

Design a parametric resonator filter with peaks at $f_0 = 1000$ Hz, $f_1 = 3000$ Hz and 3-dB widths $\Delta f_0 = 50$ Hz, $\Delta f_1 = 150$ Hz for operation at a sampling rate of $f_s = 8000$ Hz.

(a) Plot the magnitude-squared response of the filter. Indicate on your plot the location of the peaks and 3-dB widths meeting the design specifications.

(b) Plot the pole/zero pattern

Hint: Design separate resonator filters and add the transfer functions to get a single filter. If needed, you may adjust the parameters to ensure unity gain at the peaks.

Prob. 2

Consider the companion audio file, signal1.wav, consisting of a speech signal mixed with two interfering tones.

(a) Plot the magnitude spectrum of the signal (dB vs. Hz) and estimate the frequencies (in Hz and rads/sample) of the two interfering tones. For convenience, each frequency is known to be a multiple of 100 Hz.

(b) Design a filter with notches at the interfering frequencies. The filter should have a zero response at the interfering frequencies and (to the best extent possible) have a 0 dB magnitude response across the remaining freqeuncies. List the filter coefficients and plot the magnitude response (dB vs. Hz), pole-zero pattern, and impulse response of the filter.

(c) Filter the speech signal. Be sure to listen to the input and output signals. Plot the magnitude spectrum of the output signal (dB vs. Hz), compare to (a) and comment.

(d) Save the output signal as .wav file and include with the submitted codes.

Prob. 3

Use the code skeleton given on the next page to implement the grdplot.m tool. This function plots the group delay of a digital filter.

(a) Use the equations given in the *DSP Software Toolkit* book to explain exactly how the code calculates group delay. Your explanation should not simply translate MATLAB to English but rather give a detailed explanation on how the calculation is implemented.

Plot the phase response [$\angle H$ (rads) vs. ω (rads/sample)] and group delay [GRD (samples) vs. ω (rads/sample)] of the following causal filters. Based on your plots, indicate whether the filter has a *linear* phase response.

(b) $H_1(z) = -0.0078 + 0.0645z^{-1} + 0.4433z^{-2} + 0.4433z^{-3} + 0.0645z^{-4} - 0.0078z^{-5}$

(c) $H_2(z) = \dfrac{0.2929 + 0.5858z^{-1} + 0.2929z^{-2}}{1.0000 + 0.1716z^{-2}}$

(d) $H_3(z) = -0.2037z^{-1} + 0.5926z^{-2} - 0.4074z^{-3}$

Code skeleton for grdplot.m

```
%----------------------
% Calculate Group Delay
%----------------------
N = 1024;
w = [0:2*pi/N:2*pi-2*pi/N]';

jdB = dtft([0:length(b)-1]'.*b,w);
B = dtft(b,w);
tau = real(jdB ./ B);

if (length(a) ~= 1) % IIR Case
   jdA = dtft([0:length(a)-1]'.*a,w);
   A = dtft(a,w);
   tau = tau - real(jdA ./ A);
end;

%-----------------
% Plot Group Delay
%-----------------
if (fs == 2*pi)
   f = [0:2*pi/N:2*pi-2*pi/N]';
elseif (fs == 1)
   f = [0:1/N:1-1/N]';
else
   f = [0:fs/N:fs-fs/N]';
end;

if (any(imag(b))|any(imag(a))) % complex case
   plot(f,tau,'k');
else  % b,a are real
   plot(f(1:N/2),tau(1:N/2),'k');
end;

if (fs == 2*pi)
   xlabel('w (radians/sample)');
elseif (fs == 1)
   xlabel('Normalized frequency (f/fs)');
else
   xlabel('f (Hz)');
end;
ylabel('Group Delay (samples)');
grid;
```