New Mexico State University
Klipsch School of Electrical Engineering

EE395 - Introduction to Digital Signal Processing

Fall 2012
Exam #1 Part 1

Name: ____________________

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

Prob. 1

(a) Consider a continuous-time signal, $x(t)$ and the corresponding, *ideally*-sampled signal

$$\hat{x}(t) \quad = \quad \sum_{n=-\infty}^{\infty} x(nT)\delta(t-nT) \tag{1.1}$$

where T is the sample period in seconds. We model a *practically*-sampled signal as

$$x_{\text{flat}}(t) \quad = \quad \hat{x}(t) * p(t) \tag{1.2}$$

where $*$ denotes convolution and

$$p(t) \quad = \quad \begin{cases} 1, & 0 \leq t \leq \tau \\ 0, & \text{otherwise} \end{cases} \tag{1.3}$$

is the *flat-top* impulse response of an analog filter. The practical model takes into account that in an actual circuit, each sample must be held constant for a short period of time, $\tau < T$.

(a) Show that the model results in[1]

$$x_{\text{flat}}(t) \quad = \quad \sum_{n=-\infty}^{\infty} x(nT)p(t-nT). \tag{1.4}$$

[1] Please show the steps in your derivation.

Prob. 1 (cont.)

(b) Determine the frequency response of the filter, $P(f)$ by computing the Fourier Transform (FT) of $p(t)$.[2] Express your result in terms of sinc(). The FT is given by

$$P(f) \;=\; \int_{-\infty}^{\infty} p(t)e^{-j2\pi ft}dt. \tag{1.5}$$

(c) For $\tau = 1$ ms and $f_s = 100$ Hz, graph $|P(f)|$ vs. f. Note on your graph, the actual value of $|P(f)|$ at $f = f_s/2$ and $f = f_s$.

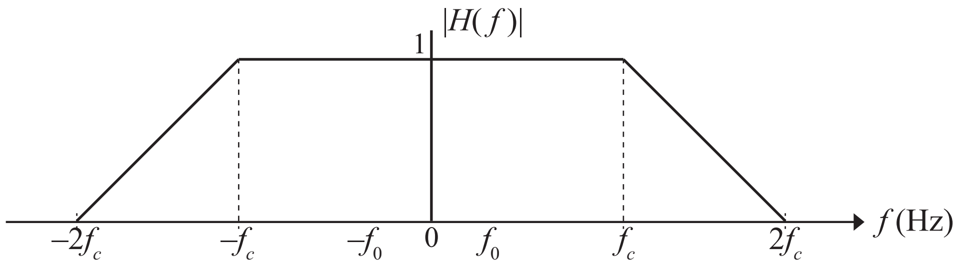
[2] Please show the steps in your derivation.

Prob. 2

Consider the continuous-to-discrete (C/D) time converter on p. 7 Figure 1.3.5. Suppose the input signal is given by

$$x_{\text{in}}(t) \quad = \quad \cos(2\pi f_0 t) \tag{2.1}$$

and the magnitude response of a practical, analog, anti-aliasing (lowpass) prefilter is shown below. Assume the passband edge frequency, $f_c = 400$ Hz.



(a) Determine the minimum value for f_s so that if $f_0 = 100$ Hz, there is no aliasing in the discrete-time signal.

(b) Determine the minimum value for f_s so that if $f_0 = 200$ Hz, the aliased spectral components within the frequency range of interest are suppressed by more than 50% relative to the signal components.

(c) Determine the minimum value for f_s so that if $f_0 = 300$ Hz, the aliased spectral components within the frequency range of interest are suppressed by more than 1% relative to the signal components.

Prob. 3

(a) Consider a digital filter with impulse response

$$h(n) = \delta(n) - 2\delta(n-1) + 2\delta(n-3) - \delta(n-4). \quad (3.1)$$

Determine the following:

(i) Input/Output (I/O) difference equation

(ii) Direct form realization with internal states indicated

(iii) Sample-by-sample processing algorithm

Prob. 3 (cont.)

(b) Consider a digital filter with I/O difference equation

$$y(n) = -x(n-1) + 2x(n-3) + 2x(n-5) - 3x(n-7). \tag{3.2}$$

Determine the following:

(i) Causal impulse response, $h(n)$ $n \geq 0$

(ii) Direct form realization with internal states indicated

(iii) Sample-by-sample processing algorithm

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Exam #1 Part 2

(Solution due in GA160G 10:30am Fri., Sep. 28, 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		/ 16 points
Prob. 2		/ 17 points
Prob. 3		/ 17 points
Total		/ 50 points

Directions

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, Sep. 28, 2012**.

Lab Hours, Office Hours, and Appointments

Extended office hours will be held on Thu., Sep. 27, 2:00–4:00pm. 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.

Prob. 1

Reconsider p. 12 Example 1.4.3. Use the singen tool to generate five samples of each of the five signals. Show that their sampled values are the same by listing them in a 5×5 table.

Prob. 2

A uniform quantizer is optimal, in an Signal-to-(quantization) Noise-Ratio (SNR) sense, when the input signal has samples which are uniformly distributed. For signals whose samples are not uniformly distributed, like speech signals, the uniform quantizer is not optimal.

Quantizer SNR performance for speech signals can be improved by *com*pressing[1] the samples prior to quantization and then ex*panding* afterward. Compression of the speech samples forces the distribution to be more uniform. The *companding* process leads to the μ-law quantizer. For more information, see

http://en.wikipedia.org/wiki/M-law_algorithm

(a) Requantize the TIMIT speech signal from Homework #2 with 3, 5, and 8 bits per sample (three separate signals). Measure the SNR for each quantized signal and list your values in a table. The SNR should be less than the 6 dB/bit rule since speech samples are not uniformly distributed.

(b) Implement the `uni2mu.m` and `mu2uni.m` tools from *DSP Software Toolkit*.[2] Compress the TIMIT samples using `uni2mu.m` with $\mu = 100$. Requantize the compressed TIMIT speech signal with 3, 5, and 8 bits per sample (three separate signals) and then expand each using `mu2uni.m`. Measure the SNR for each μ-law quantized signal, list your values in a table.

(c) Compare your SNR results in (b) with (a) and comment. In addition to comments, include plots of the sample distributions before and after compression.

[1] Compression in this case means *dynamic range* compression, i.e. large-amplitude samples become smaller and small amplitude samples become larger.

[2] Note that these tools have been recently updated in the *DSP Software Toolkit*. Please download at http://www.ece.nmsu.edu/~pdeleon/Teaching/EE395/DSP_Toolkit.pdf

Prob. 3

Reconsider the 7-point Moving Average (MA) filter from Homework #3, Problem #5. Let

$$x(n) \quad = \quad \cos\left(2\pi\frac{f_0}{f_s}n\right), \quad 0 \leq n \leq 49 \tag{3.1}$$

where $f_s = 200$ Hz be the input signal. For each of the following frequencies, f_0, filter the input signal. Plot, separately, the input and output signals and measure the frequency attenuation of the filter on the steady-state output signal.

(a) $f_0 = 0.5 \times f_s/7$

(b) $f_0 = f_s/7$

(c) $f_0 = 1.5 \times f_s/7$

(d) $f_0 = 2 \times f_s/7$

(e) $f_0 = 2.5 \times f_s/7$

(f) $f_0 = 3 \times f_s/7$