New Mexico State University
Klipsch School of Electrical Engineering

EE395 - Introduction to Digital Signal Processing

Fall 2010
Exam #2 Part 1

Name: _____

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

Prob. 1

Consider the LCCDE for a digital filter

$$\begin{aligned} y(n) &= 0.13x(n) + 0.40x(n-1) + 0.40x(n-2) + 0.13x(n-3) + \\ &\quad 0.33y(n-1) - 0.60y(n-2) + 0.20y(n-3). \end{aligned}$$

Coefficient assignments follow the textbook p. 233 (6-21).

(a) Determine the transfer function, $H(z)$.

(b) Draw the Direct Form I filter structure.

(c) Draw the Direct Form II filter structure.

Prob. 2

The transfer function of a digital filter is given by [p. 235 (6-25)]

$$H(z) = \frac{b(0) + b(1)z^{-1} + \cdots + b(N)z^{-N}}{1 - a(1)z^{-1} - \cdots - a(M)z^{-M}}$$

whereas in factored form we have [z_k denotes zeros and p_k denotes poles of $H(z)$]

$$H(z) = \frac{(1 - z_1 z^{-1})(1 - z_2 z^{-1}) \ldots (1 - z_N z^{-1})}{(1 - p_1 z^{-1})(1 - p_2 z^{-1}) \ldots (1 - p_M z^{-1})}.$$

Assume we store the filter coefficients in column vectors as

$$\begin{aligned} \mathbf{b} &= [b(0),\ b(1), \ldots,\ b(N)]^T \\ \mathbf{a} &= [1, -a(1), \ldots, -a(M)]^T . \end{aligned}$$

For the following coefficient vectors, determine the transfer function $H(z)$ and whether the filter has a finite-length impulse response (FIR) or infinite-length impulse response (IIR).

Note: Cancellation of factors in $H(z)$ may influence your decision.

(a) [FIR / IIR] $\mathbf{b} = [0, 1, 1/2]^T$ and $\mathbf{a} = [1]^T$

$H(z) =$

(b) [FIR / IIR] $\mathbf{b} = [1]^T$ and $\mathbf{a} = [1, -1/2]^T$

$H(z) =$

(c) [FIR / IIR] $\mathbf{b} = [1, 0, -1/4]^T$ and $\mathbf{a} = [1, 1/2]^T$

$H(z) =$

(d) [FIR / IIR] $\mathbf{b} = [1, -1/2]^T$ and $\mathbf{a} = [1, 0, 1/4]^T$

$H(z) =$

Prob. 3

Figures 3.1 and 3.2 on the next two pages contain magnitude and phase responses of eight digital filters designed as follows:

- Design: rectangular-window sinc function, hamming-window sinc function, Butterworth, or Chebyshev
- Filter order: sinc function, $N = 32$ or $N = 128$; Butterworth or Chebyshev $N = M = 3$ or $N = M = 19$.

Four designs and two possible orders for each design yields eight filters. These filters are represented exactly *once* in the figures.

Fill the table below. Feel free to tear off the next two pages to expedite your answer.

Table 1:

Figures	Design	N	FIR or IIR?	Linear or Nonlinear Phase Response?
3.1(a)(b)				
3.1(c)(d)				
3.1(e)(f)				
3.1(g)(h)				
3.2(a)(b)				
3.2(c)(d)				
3.2(e)(f)				
3.2(g)(h)				

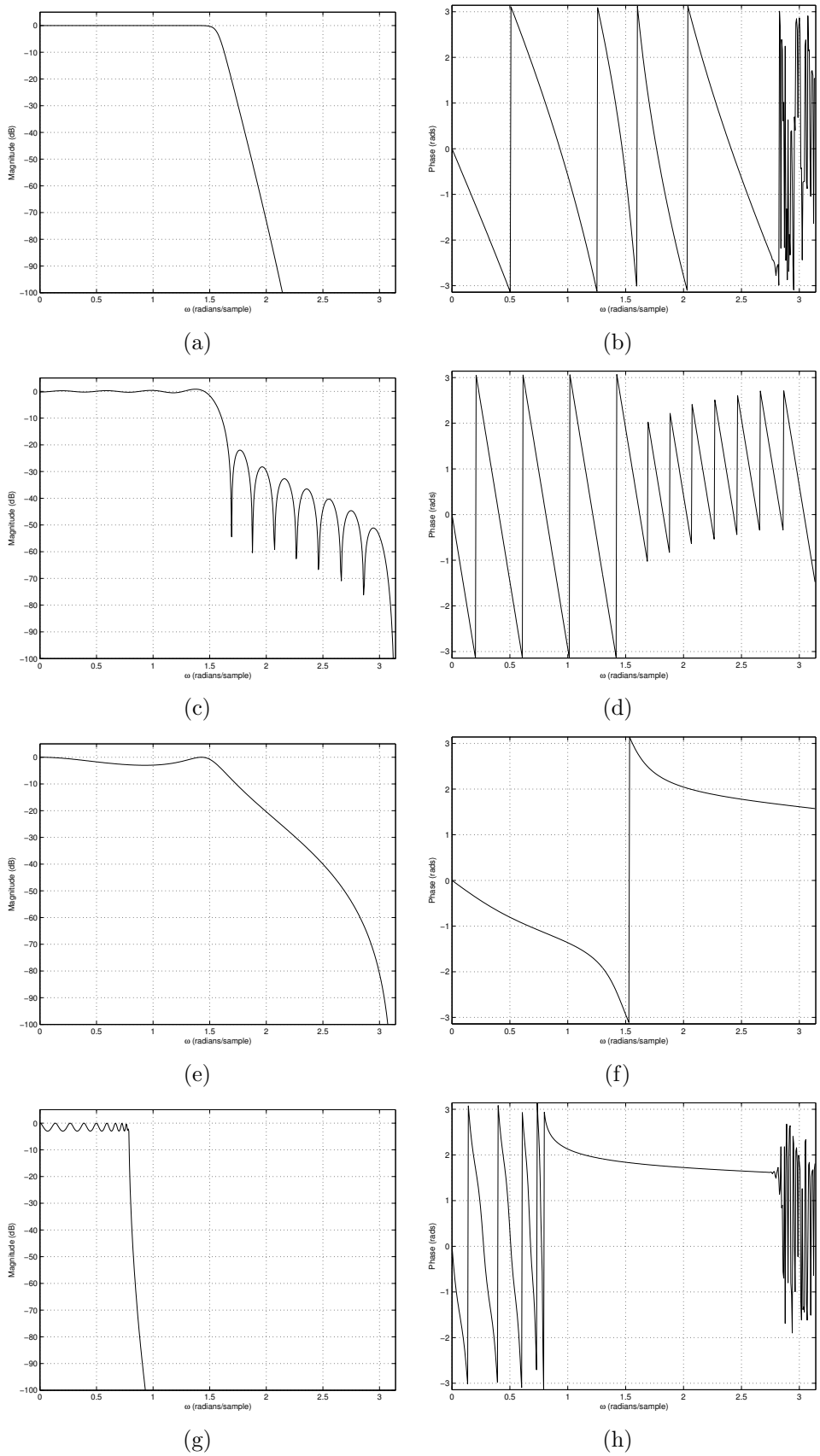

Figure 3.1: Magnitude and phase responses for Problem 3.

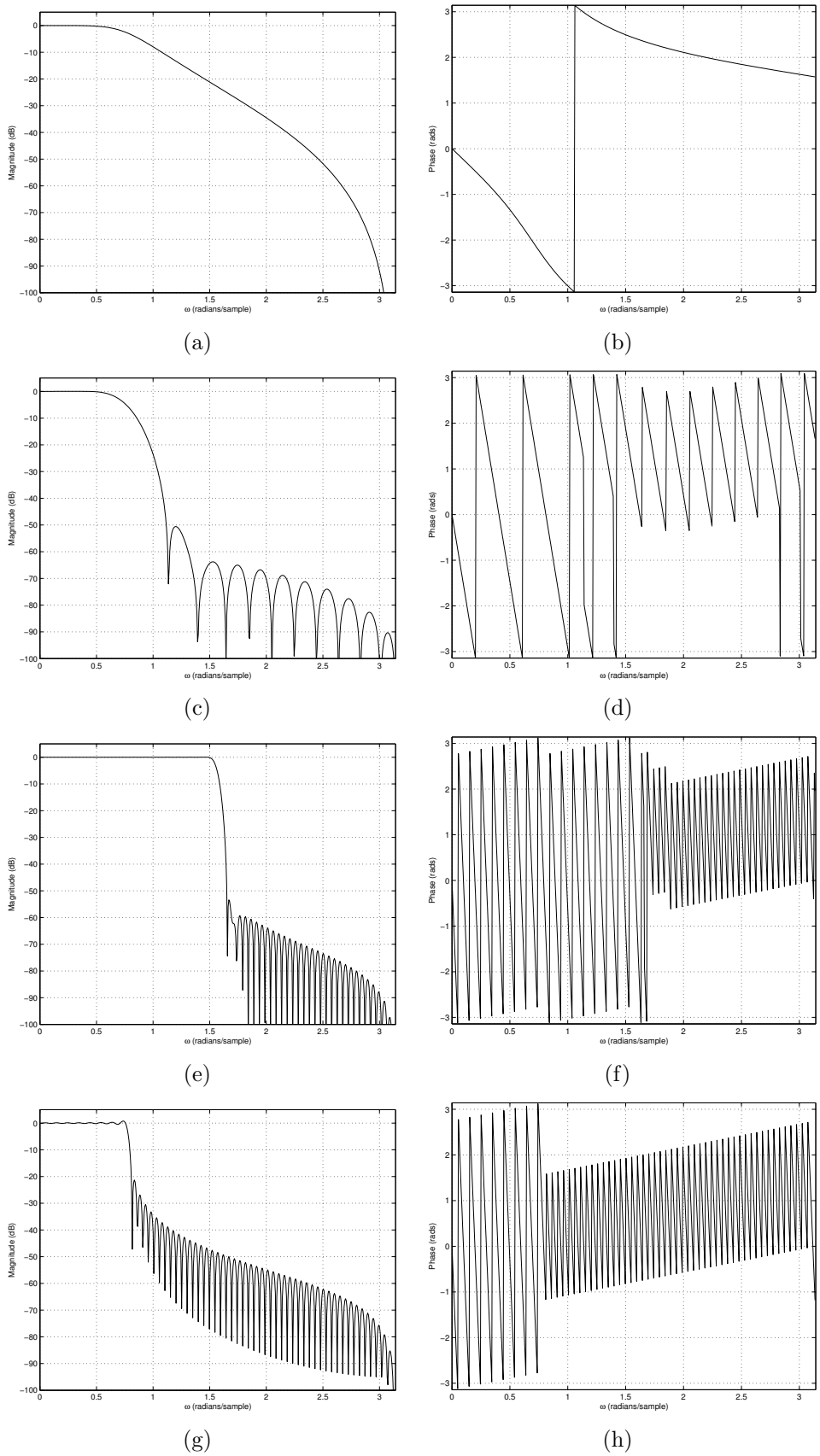

Figure 3.2: Magnitude and phase responses for Problem 3.

New Mexico State University
Klipsch School of Electrical Engineering

EE395 - Introduction to Digital Signal Processing

Fall 2010
Exam #2 Part 2

(Solution due in GA160G 10:30am Fri., Oct. 29, 2010)

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

Signature: _____ Date: _____

Prob. 1	/ 25 points
Prob. 2	/ 25 points
Total	/ 50 points

Directions

For each problem, please write a MATLAB code which produces the required plots. Name your main codes prob1.m and prob2.m. Please submit the *signed* cover sheet of this exam and a hardcopy of your solutions (plots and main codes). In addition, please send (before the due date) an email to `pdeleon@nmsu.edu` with a .zip file archive containing the two main codes and a directory called "DSP_toolkit" containing your tools. You will receive an email confirmation upon receipt of the .zip file.

Required Signals

Please download

http://www.ece.nmsu.edu/~pdeleon/Teaching/EE395/Exam2Part2.zip

which contains signal(s) required for this exam.

Assistance

- You are free to email any questions to Prof. De Leon during the project period. Email may include a request to examine code.
- Extended office hours will be held on Thu., Oct. 28, 9:00–11:00am and 3:00–5:00pm.

Prob. 1

Most digital audio players include an *equalizer* that allows a user to customize the sound spectrum (using digital filters!) according to their preferences. The 10-band equalizer for Apple's iTunes (Rock setting) is shown in Fig. 1.1. We believe, the frequency value (beneath the slider) indicates the center frequency, f_0 (in Hz) and the slider value is the gain, G (in dB) at f_0. Assume $f_s = 48000$ Hz.

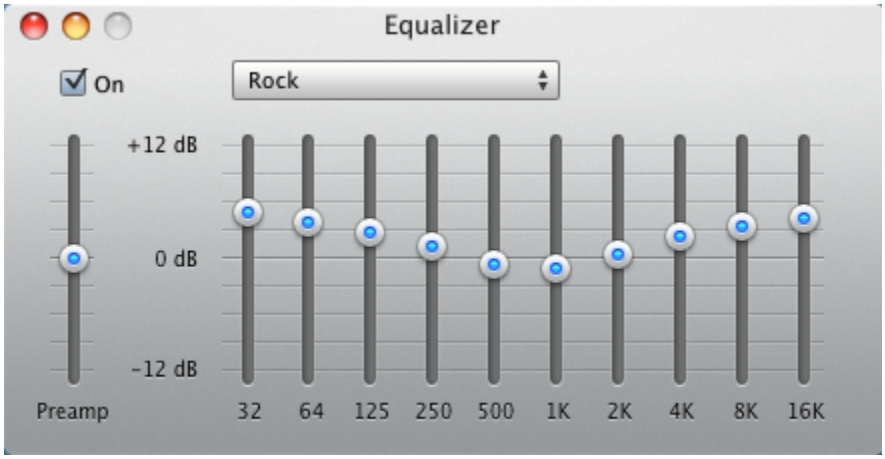

Figure 1.1: iTunes equalizer

A *parametric equalizer* consists of a bank of digital filters whose parameters—gain, center frequency, and bandwidth—can be varied. The parameters are the reference gain G_0, center frequency ω_0 (rads/sample), filter gain G at ω_0, and the desired width $\Delta\omega$ at an appropriate gain G_B as illustrated in Fig. 1.2.

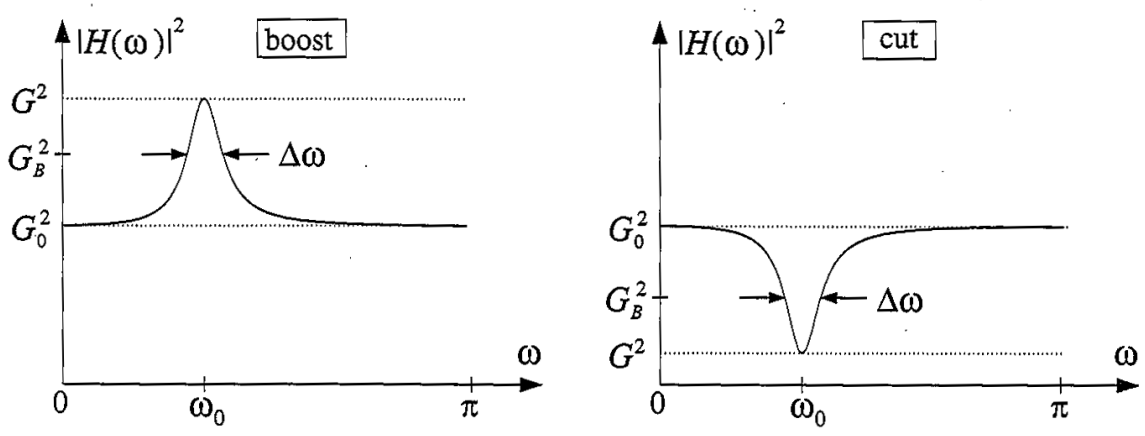

Figure 1.2: Parametric equalizers for frequency boost or cut.

The transfer function for a parametric equalization filter is [1]

$$H(z) = \frac{\left(\frac{G_0+G\beta}{1+\beta}\right) - 2\left(\frac{G_0\cos\omega_0}{1+\beta}\right)z^{-1} + \left(\frac{G_0-G\beta}{1+\beta}\right)z^{-2}}{1 - 2\left(\frac{\cos\omega_0}{1+\beta}\right)z^{-1} + \left(\frac{1-\beta}{1+\beta}\right)z^{-2}} \tag{1.1}$$

where

$$\beta = \sqrt{\frac{G_B^2 - G_0^2}{G^2 - G_B^2}}\tan\left(\frac{\Delta\omega}{2}\right). \tag{1.2}$$

(a) Design parametric equalizer filters that approximate the iTunes filters centered at 1000, 2000, 4000, and 8000 Hz (estimate the filter gains from Fig. 1.1). Assume $G_0 = 1$ and

$$G_B = \sqrt{\frac{G_0^2 + G^2}{2}}. \tag{1.3}$$

Be sure to convert filter gains in dB to normal units for use in the above formula:

$$g = 10^{g_{\text{dB}}/20}. \tag{1.4}$$

You may need to experiment with $\Delta\omega$ for each filter in order to get satisfactory performance at the crossover frequencies. On a *single* figure, plot the magnitude responses of your filters in dB. Be sure to provide at least one plot which zooms in on 0 dB.

(b) Use the code below to process signal1.wav (from the downloaded .zip file) which uses a serial arrangement of your filters. Comment on the amplitudes of the sinusoids, i.e. frequencies before and after filtering. Are they consistent with your filter gains?

```
[x,fs,bits] = wavread('signal2.wav');
y1 = filter(b1,a1,x); % output of filter 1, coefs in vectors b1 and a1
y2 = filter(b2,a2,y1); % output of filter 2
y3 = filter(b3,a3,y2); % output of filter 3
y = filter(b4,a4,y3); % output of filter 4
figure(1);plotcsig(x,fs);
figure(2);plotcsig(y,fs);
```

[1] S. Orfanidis, *Introduction to Signal Processing*, Prentice-Hall, 1996.

Prob. 2

In this problem, we will design a "notch" filter in order to remove a narrow-band interference from a wide-band signal. To begin, listen to signal2.wav from the downloaded .zip file.

(a) Code the welch2.m and periodogram_plot.m tools in the DSP Toolkit which implements Welch's method for estimation of the power spectrum. This technique averages the power spectra from a sequence of short, overlapped, consecutive signal segments.

(b) Compute and plot the periodogram (in units of dB and Hz) of the signal using a 1024-point Hamming window and an overlap factor of 256 samples (25% overlap).

(c) From the periodogram plot, determine the frequency of the interference, f_{noise} in Hz and ω_{noise} in rads/sample.

(d) Design a simple notch filter to remove the narrowband interference by creating a transfer function $H(z)$ with two zeros located at precisely $e^{\pm j\omega_{\text{noise}}}$, i.e.

$$\begin{aligned} H(z) &= (1 - e^{j\omega_{\text{noise}}} z^{-1})(1 - e^{-j\omega_{\text{noise}}} z^{-1}) \\ &= 1 - 2\cos(\omega_{\text{noise}}) z^{-1} + z^{-2}. \end{aligned} \tag{2.1}$$

Plot the pole-zero pattern and magnitude response (in units of dB and Hz) of your notch filter.

(e) Filter the signal and confirm the elimination of the narrow-band interference. Use MATLAB's filter.m if your fltr.m runs slow.

(BONUS +5) Sharpen your notch by placing two poles at precisely $re^{\pm j\omega_{\text{noise}}}$ where you are free to choose $0 < r < 1$. Plot the polezero pattern and magnitude response of your sharpened notch filter.