New Mexico State University
Klipsch School of Electrical Engineering

EE590 - Digital Speech Processing
Fall 2003 - Exam #2

Name: ______________________

"The attached solution is due entirely to my own, individual efforts. I have not discussed this exam 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
Prob. 3	/ 25 points
Prob. 4	/ 25 points
Total	/ 100 points

Directions

- Your solution to this exam is due on or before **5:00pm Thursday, Dec. 11, 2003**. A printed solution is strongly preferred but an electronic solution in PDF format is also accepted. However, no effort will be made to decode, decipher, or assemble a "buggy" electronic solution.
- You are free to use any and all resources available to you to prepare solutions to this exam. These resources include textbooks, class notes, computers, MATLAB, and the instructor of this course. These resources do *not* include the Internet and other students. Any codes should be attached at the end of the solution.
- During the period of Dec. 8 – 11, Prof. De Leon will have open office hours 8:00am – 5:00pm and students are encouraged to come by and discuss the problems. Students are also free to call (646-DSP1) or email (pdeleon@nmsu.edu). Some "live" help may also be available through following instant messenging systems: pdeleon2 @ AOL, pldeleon @ MSN, pldeleon @ Yahoo. You will need to send a prior email with your instant messenging address so that you will not be blocked.

Prob. 1

(Text, Prob. 12.9) Consider a signal $x[n]$, [sample signal shown in Figure 1.1(a)] that takes on values according to the pdf given in Figure 1.1(b).

see text, p. 655, Figure 12.32

Figure 1.1: Quantization conditions for Prob. 1: (a) sample function; (b) pdf of $x[n]$; (c) uniform quantizer; (d) nonuniform quantizer.

(a) Derive the signal-to-noise ratio (SNR) defined as

$$\text{SNR} = \frac{\sigma_x^2}{\sigma_e^2} \tag{1.1}$$

for a 2-bit quantizer where the reconstruction levels are uniformly-spaced over the interval $[0, 1]$, as shown in Figure 1.1(c). Assume the pdf of the quantization noise is uniform. Note that you should first find the range of the quantization error as a function of quantization step size Δ.

(b) Now suppose we design a nonuniform quantizer that is illustrated in Figure 1.1(d). Derive the SNR for this quantizer. Assume the pdf of the quantizer error $e[n]$ is uniform for $x[n]$ in the interval $[0, 1/2]$, and also uniform for $x[n]$ in the interval $[1/2, 1]$, but within each interval a different quantization step size is applied, as shown. *Hint*: Use the relation:

$$p_e(e[n]) = \text{Prob}\{x \in [0, 1/2]\} \cdot p_e(e[n] \mid x \in [0, 1/2]) + \tag{1.2}$$

$$\text{Prob}\{x \in [1/2, 1]\} \cdot p_e(e[n] \mid x \in [1/2, 1]) . \tag{1.3}$$

(c) Is the nonuniform quantizer of part (b) an optimal Max quantizer? Explain your reasoning.

(d) The following MATLAB code generates a test signal according to the conditions of this problem:

```
rand('state',0);
tmp1 = 0.5*rand(1000,1);        % 1000 random numbers from 0 to 0.5
tmp2 = (0.5*rand(7000,1))+0.5; % 7000 random numbers from 0.5 to 1.0
tmp = [tmp1;tmp2];              % "ordered" random numbers
index = randperm(8000);         % random permutations of integers from 1 to 8000
x = tmp(index);                 % "unordered" random numbers
```

Plot the first 100 samples and the pdf of the test signal. Are your plots similar to Figures 1.1(a) and (b) (YES/NO)?

(e) Quantize the test signal according to Figure 1.1(c) and compute the SNR in normal units. How does your value compare to that in (a)?

(f) Quantize the test signal according to Figure 1.1(d) and compute the SNR in normal units. How does your value compare to that in (b)?

Prob. 2

(a) Let $x_1[n]$ and $x_2[n]$ denote two sequences and $\hat{x}_1[n]$ and $\hat{x}_2[n]$ the corresponding complex cepstra. If $x_1[n] * x_2[n] = \delta[n]$, determine the relationship between $\hat{x}_1[n]$ and $\hat{x}_2[n]$.

(b) Suppose that the complex cepstrum of $y[n]$ is $\hat{y}[n] = \hat{s}[n] + 2\delta[n]$. Determine $y[n]$ in terms of $s[n]$.

(c) Determine the complex cepstrum of $x[n] = 2\delta[n] - 2\delta[n-1] + 0.5\delta[n-2]$.

(d) (Similar to text, p. 266, example 6.6) Let

$$H(z) \quad = \quad \frac{(1 - bz^{-1})(1 - b^* z^{-1})}{(1 - cz^{-1})(1 - c^* z^{-1})} \tag{2.1}$$

where $b = 1.01e^{j0.12\pi}$, $c = -0.99e^{j0.12\pi}$, and the ROC is the set $|z| > 0.99$. This $H(z)$ is causal unlike that in the text. Let

$$p[n] \quad = \quad \beta^n \sum_{k=0}^{\infty} \delta[n - kP]. \tag{2.2}$$

Assuming a sample rate of $f_s = 10000$ choose P so that peaks occur every 10ms as in text, p. 267, Figure 6.9. Plot the complex cepstrum (use MATLAB's `cceps` function), $\hat{x}[n]$ where $x[n] = h[n] * p[n]$ and thus $\hat{x}[n] = \hat{h}[n] + \hat{p}[n]$.

Note: In order for your complex cepstum plot to resemble that in Figure 6.9, you will have the reorder the lower and upper halves of the cepstrum (this is equivalent to plotting the range $[0, 2\pi]$ or $[-\pi, \pi]$?).

(BONUS +10points) Exactly recreate Figure 6.6.

Prob. 3

Let $x[n]$ be the "clean" signal from your `Name.wav` and let $v[n]$ be an additive, white Gaussian noise (AWGN). Using the `add_noise.m` code, create a "noisy" speech signal, $y[n]$ with a 0dB SNR.

(a) Plot periodograms of both the clean speech, $S_x(\omega)$ and noisy speech $S_y(\omega)$.

(b) Plot the first 15 lags of the unbiased correlation sequence, $\mathbf{r}$ of $x[n]$.

(c) Using $\mathbf{r}$ to build the Toeplitz matrix, $\mathbf{R}$ (16×16), plot the elements of the eigenvector, $\mathbf{w}$ corresponding to the maximum eigenvalue. Scale the coefficients (if necessary) so that the filter noise gain is 1. The filter $\mathbf{w}$ is the *eigenfilter* for $x[n]$.

(d) Plot the magnitude response (in dB) of the eigenfilter as a function of f in Hertz. Based on the plots in (a) and (d), why does this filter make sense?

(e) Apply the eigenfilter to $y[n]$ and plot the periodogram of the enhanced speech. Listen to the enhanced speech. Comment.

Prob. 4

In this problem we will develop a short code to perform a time-scale modification of a speech signal. To begin, code the `OLA.M` tool on the next page.

(a) Using your `Digits.wav`, plot the signal (use `plotcsig2`) and the spectrogram (STFT).

(b) Time-scale expand the signal by a factor of 2, i.e. create a modified STFT by simply repeating each frame in the STFT from (a). Synthesize the time-scale expanded signal from the modified STFT. Plot the synthesized signal and its spectrogram.

(c) Listen to the modified signal. The articulation rate shold be S-L-O-W (with little distortion) while maintaining the pitch. Comment on the plots in (a) and (b).

OLA.M

Purpose

This function synthesizes a real, time-domain signal from a short-time Fourier transform (STFT) using the overlap-add (OLA) method.

Input

X, STFT computed using a triangular window (50% overlap). Spectra are stored as columns.

Output

x, synthesized signal

Algorithm

Apply an inverse DFT to each column of **X**, overlapping the resulting signals by 50% and adding.

Notes

1. The beginning and end of **x** may not be accurate due to edge effects.
2. Due to numerical issues, there may be a small imaginary part in **x**. In this case, simply return real(x).

Reference

T. Quatieri, *Discrete-Time Speech Signal Processing*, Prentice-Hall: Upper Saddle River, N.J., 2001.

Example

```
>>rand('state',0);
>>N = 16; % window length
>>x = rand(4*N,1); % four windows of noise
>>x(1) = j*10^(-10); % add small imag. to get complete spectrum from specgram
>>w = triang(N); % build window
>>X = specgram(x,N,1,w,N/2); % compute STFT of noise
>>y = ola(X); % synthesize noise signal from STFT
>>e = real(x-y); % compute difference
>>plot(e);ylabel('e[n]');xlabel('n');grid; % visualize accuracy
```

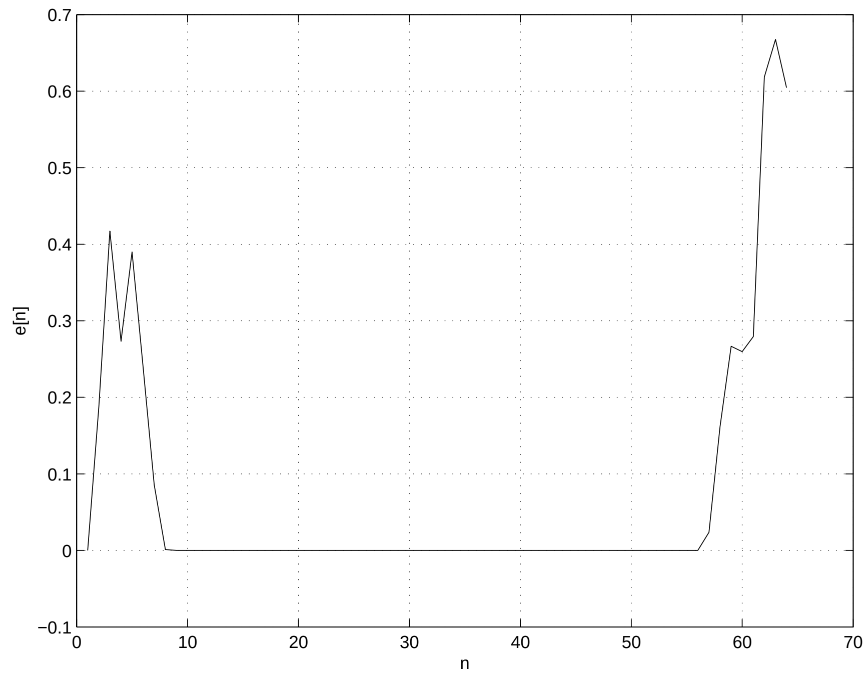

Figure 4.1: OLA. Error is due to edge effects