Cairo University
Faculty of Engineering
Biomedical Engineering Department

May 2007
Time: 2 Hours
Open Book/Notes

BIOMEDICAL SIGNAL PROCESSING FINAL EXAM

SOLVE AS MUCH AS YOU CAN – MAXIMUM GRADE: 60 POINTS

QUESTION GROUP I: SOLVE THE FOLLOWING PROBLEMS:

1. [5 points] In a computer-aided diagnosis problem applied to digital mammography, it is desired to take a region-of-interest (ROI) within the image and analyze it to generate descriptive features that can be used for classification. Assume that you have a 32x32 region and that you will rearrange the points in this region to fill a 1D vector of length 1024 by concatenating the rows in that ROI. Design a digital signal processing system that enables the extraction of useful features from such vectors.
2. [5 points] In a bioinformatics signal processing system, the protein sequences in a genome are composed of an array of 4 types of proteins denoted as A, C, G, and T proteins. Their distribution within the sequence determines the characteristics of the genome and reveals important information that can be used in many areas. In one application of bioinformatics, it is desired to take a small strand (short sequence taken from a longer genome sequence) taken from a crime scene and match it to the genetic records from possible suspects. If such matching is determined solely based on the largest number of individual matches within the sequence, design a digital signal processing that enables this to be done.
3. [5 points] In brain-computer interfacing, the EEG data record comes in the form of an array of 30000 points sampled over 30 seconds at a sampling rate of 1000 samples/second. If the frequency range of the sampled EEG is from 0.01Hz to 40Hz, design a digital signal processing system that filters the data to this range only.
4. [5 points] In Doppler ultrasound imaging, a continuous-wave 10 MHz ultrasound is sent and results in a ±10 kHz of Doppler shift. The Doppler-shifted data are sampled using one channel at a sampling frequency of 50 MHz. Design a digital signal processing system that implements a Hilbert transformation to these data and determine the largest reduction in sampling possible after the Hilbert transform is performed on the data.
5. [5 points] In infrared spectroscopy system, it is desired to calculate an accurate power spectrum for a biomedical signal. Assume that the sampling rate is 1 MHz and the number of samples acquired is 100000 samples. Design an algorithm to calculate the power spectrum of the data given that the desired frequency domain resolution is 1 kHz. Make sure that the SNR of the resultant power spectrum is optimal.
6. [5 points] To design a digital FIR band-pass filter of cut-off frequencies of 60 and 200Hz (given the sampling frequency of 40Hz), it is desired to have a 4 point filter which satisfy the following constraints: a) H(0 Hz)=0, H(80 Hz)=1, H(180 Hz)=1, H(220 Hz)=0. Select a method to solve this problem and write down the equations to be solved to calculate the filter coefficients in an optimal manner.

7. [5 points] In an embedded DSP system, a DSP processor that allows real-time processing of data. The DSP system computes the spectrogram for a biomedical signal under the following conditions: sampling rate: 10 kHz, window size= 128, number of windows to compute per second=100, a hamming window is used in each case and averaging is not used. Design a suitable digital signal processing method to do that and estimate the processing power required (in terms of an order of computations/second).

<u>Question Group II:</u>
<u>Label the following Statements as either True or False and explain your answer:</u>

<u>(Read Carefully: Grading policy for each: correct answer and explanation = 3 points, incorrect answer and/or incorrect explanation= –3, no answer = 0)</u>

1. The Hilbert transform can be used to compute the complex analytic signal given only one of its components (e.g., the real part only).
2. The theory of infinite impulse response digital filters has roots in analog filter design methods.
3. Windowing is mainly used to reduce the length of the data to be processed by FFT and consequently reduces the required computation time.
4. The discrete Fourier transform is an approximation for discrete time Fourier transform.
5. It is possible to completely recover an analog signal from its digital samples if the sampling rate was higher than the bandwidth of the signal.
6. Sampling rate required for a given signal is usually much higher than the sampling rate required for the quadrature version of the same signal.
7. Spectral leakage has the same source and meaning as Gibbs ringing.
8. Power spectrum estimation is best used for signals with time-varying frequency content.
9. Stability in digital filters means correct computations without overflow or underflow.
10. It is possible to optimize both the main lobe width and side lobe magnitude simultaneously using different windows.
11. Linear prediction coefficients can be calculated using the Fourier transformation.
12. Spectrograms are two dimensional images with a horizontal axis of time and vertical axis of frequency.
13. The only Fourier transform property used in DTFT is the time-domain sampling resulting in a periodic Fourier transform.
14. Linear convolution cannot be computed using the DFT.
15. Real signals always have a symmetric Fourier transformation magnitude.

Best of Luck!