BIOMEDICAL ENGINEERING DEPARTMENT **SEPTEMBER 2002**
CAIRO UNIVERSITY **OPEN BOOK**

BIOMEDICAL DIGITAL SIGNAL PROCESSING FINAL EXAM

Time Allowed: 2 Hours – Maximum Number of Points: 60 – Solve All Questions

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PART I: Design Problems

1. [4 Points] Assume the received signal to have a bandwidth of 1 MHz around a center frequency of 10 MHz. Design a suitable and efficient sampling schemes using both a single channel or quadrature sampling.

2. [4 Points] Describe a methodology to estimate an accurate power spectrum of a signal under the following constraints:

- Sampling rate of signal is 1kHz
- Sampling window length is 20 sec.
- Desired resolution in the frequency domain is 0.2 Hz

3. [4 Points] Design a suitable digital signal processor to compute the spectrogram given the following specifications:

- The data are sampled at a rate of 3000 samples/second.
- Spectrogram time-domain window is 256-point windows (i.e., the same as the length in spectral direction is 256 points).
- Number of windows processed per second is 30

4. [4 Points] Describe how to implement a technique to extract a model for a given biological signal. Assume all the missing information.

5. [4 Points] Describe how to design an optimal filter in the following cases:

- If the noise is stationary and known to be white Gaussian noise.
- If the noise is known to be white Gaussian noise but with time-varying parameters.

6. [4 Points] Describe how to design an optimal FIR filter to satisfy the following frequency domain response characteristics:

Frequency	0.15π	0.25π	0.6π	0.8π
Response	1	0.9	0.2	0.1

PART II: Miscellaneous Problems

7. [8 Points] It is desired to transform the following analog filter to a digital filter:

$$H(s) = \frac{s+2}{s^2+3s+3}$$

Assume the sampling rate to be 100Hz. Calculate the digital filter transfer function.

8. [4 Points] Given an FIR filter with h(-2)=1, h(1)=0,h(0)=0,h(1)=-1, calculate the output of filtering a periodic sequence x with period described as: x(0)=1, x(1)=2, x(2)=3, x(3)=4.

9. [5 Points] For each of the following systems, determine whether or not the system is linear:

(a) $y(n) = T[x(n)] = \sum_{k=0}^{k=n} (k+1) \cdot x(k)$

(b) $y(n) = T[x(n)] = \max_{n+1>k>n-1} \{x(k)\}$

(c) $y(n) = T[x(n)] = 2x(n-1) + 3x(n-2) + 2x(0)$

(d) $y(n) = T[x(n)] = 10(x(n) + x(n+1))$

(e) $y(n) = T[x(n)] = \sum_{k=-\infty}^{\infty} g(k) \cdot x(n-k)$

10. [3 Points] Let *h(n)* denote the impulse response of a 1-D low-pass filter. Show using Fourier domain analysis that the filter *g(n)* defined as: $g(n) = (-1)^n \cdot h(n)$ represents a high-pass filter.

11. [4 Points] Consider the linear system described by the following equation:

$$y(n) = x(n) + x(n-1) - y(n-2) - y(n-3)$$

Derive the linear system transfer function. Derive the coefficients of an inverse filter that would enable the estimation of *x(n)* given *y(n).* Comment on the stability characteristics of this filter.

12. [6 Points] Obtain the inverse z-transformation for the following:

a) $H(z) = 1 + 2z^{-3} + 0.25z^{-5}$

b) $H(z) = 2/(3+2z^{-1})$, |z|>1

c) $H(z) = (2-z^{-1})/(2-3z^{-1}+z^{-2})$, |z|>1

13. [6 Points] In an embedded DSP system, a DSP processor that allows real time processing of data. The DSP system computes the spectrogram for a color Doppler system under the following conditions: window size= 256, number of windows to compute per second=256, a hamming window is used in each case and averaging is not used. Estimate a suitable processing power for this processor.

Best of Luck!