UNIVERSITY OF KWAZULU-NATAL
SHOOL OF ENGINEERING
Electrical, Electronic & Computer Engineering
MAIN EXAMINATION: JUNE 2014

ENEL4DS H1: Digital Signal Processing

Duration: 2 Hours **Marks: 80**

Examiners:

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Prof H Xu
Prof M Dlodlo (UCT)

Instructions:

1. **This is a closed book examination and no notes may be used.**
2. **Answer 4 out of the 5 questions**
3. **Formulae sheets are attached at the end of the question paper.**

Question 1

(a) What are the advantages and disadvantages of digital signal processing in comparison to analogue signal processing?

(6)

(b) Explain two methods that are used to eliminate echoes in long distance telephone networks.

(4)

(c) An analogue signal x(t) is sampled by an impulse train at the sampling $f_s = 1/T_s$ samples per second. Give expressions for the sampled signals in the time domain and frequency domains. If the highest frequency in x(t) is f_m derive the condition for zero aliasing errors.

(4)

(d) Determine the overall impulse response of the system of figure Q1(d) given that

$h_1(n) = 2\delta(n-2) - 3\delta(n+1)$,

$h_2(n) = \delta(n-1) + 2\delta(n+2)$ and

$h_3(n) = 5\delta(n-5) + 7\delta(n-3) + 2\delta(n-1) - \delta(n) + 3\delta(n+1)$.

(6)

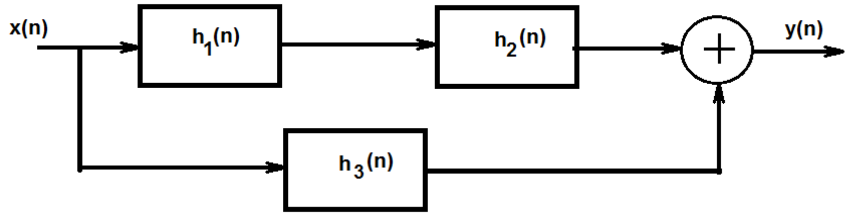

Figure Q1(d)

Question 2

(a) A moving average filter is given by $y(n)=\frac{1}{M}\sum_{k=0}^{M-1}x(n-k)$ where x(n) is the input and y(n) is the output.

(i) Show that the filter is BIBO stable.

(ii) Show whether the filter is a linear time invariant system.

(iii) Obtain the expression for the magnitude and phase frequency response of the moving average filter

(15)

(b) Show that the Fourier transform of a unit step sequence, μ(n), is given by $U(e^{j\omega})=\frac{1}{1-e^{-j\omega}}+\sum_{k=-\infty}^{\infty}\pi\delta(\omega+2\pi k)$.

(5)

Question 3

a) Explain the difference between a Discrete-Time Fourier Transform (DTFT) and a Discrete Fourier Transform (DFT).

(2)

b) Calculate the DFT of the data sequence $\{x(n)\}=(0 \quad 1 \quad 1 \quad 0)$ for $0\le n\le 3$

(5)

c) Suppose that $\{x(n)\}$, $0\le n\le N-1$ is a length-N sequence with an N– point DFT $X(k),\ 0\le k\le N-1$, show that if N is even and if $x(n)=-x(n+\frac{N}{2})$ for all n, then X(k) = 0 , for k even.

(8)

d) Compute the circular convolution of the following two sequences:

$$\{g(n)\} = \{\underset{\uparrow}{1} \quad 2 \quad 0 \quad 1\}, \qquad \{h(n)\} = \{\underset{\uparrow}{2} \quad 2 \quad 1 \quad 1\}$$

(5)

Question 4

(a) Using the z-transform methods, determine the explicit expression for the output y(n) for the causal LTI discrete time system with impulse response $h(n) = (-0.4)^n \mu(n)$ when the input is $x(n) = (0.2)^n \mu(n)$, where $\mu(n)$ is the unit step sequence.

(8)

(b) Obtain an expression for the transfer function $H(z) = \dfrac{Y(z)}{X(z)}$ for the system of Figure Q4(b). Hence, derive expressions for the magnitude and phase frequency responses for the system

(7)

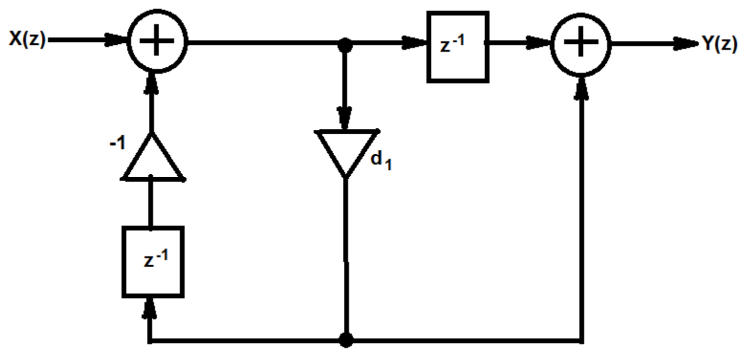

Figure Q4(b)

(c) Develop one cascade canonic realization of the transfer function given by

$$H(z) = \frac{(0.2z^{-1} + 0.6z^{-2})(3 - 2.4z^{-1})}{(2 - 3.2z^{-1} + 4.2z^{-2})(1 - 0.75z^{-1})}$$

(5)

Question 5

(a) You are to design a lowpass FIR filter using the window method to meet the following specifications:

- The passband edge frequency $\omega_p = 0.3\pi$
- The stopband edge frequency $\omega_s = 0.5\pi$
- The minimum stopband attenuation $\alpha_s = 40$ dB

Determine

(i) the order of the filter,
(ii) the impulse response of the ideal LPF h(n),
(iii) the appropriate window function, and
(iv) the first three of the truncated impulse response coefficients.

(10)

(b) A bandpass Butterworth IIR digital filter is to be designed using the bilinear z-transform method to meet the following specifications:

- The passband-edge frequencies: 2kHz and 4 kHz
- The stopband-edge frequencies: 1.8 kHz and 4.5 kHz
- The maximum passband attenuation: 0.5 dB
- The minimum stopband attenuation: 40 dB
- The sampling frequency is to be 10 kHz.

(i) Obtain the specifications of the corresponding analog bandpass filter. To obtain geometric symmetry adjust the lower passband edge frequency of the bandpass filter.

(ii) Give the specifications of the prototype lowpass filter to be used in the design.

(iii) Write a MATLAB programme that will compute coefficients of the desired digital filter starting with the prototype lowpass filter.

(10)