GENERAL SIR JOHN KOTELAWALA DEFENCE UNIVERSITY
Faculty of Engineering
Department of Electrical, Electronic and Telecommunication Engineering

BSc Engineering Degree
5th Semester Examination – May/June 2018
Intake 33-(ET/ MC)

DIGITAL SIGNAL PROCESSING
(ET3132)

Time allowed: 2 hours 24th May, 2018

ADDITIONAL MATERIAL PROVIDED

Nil

INSTRUCTIONS TO CANDIDATES

This paper contains FOUR questions on 6 pages

Answer all four questions.

THIS IS AN CLOSED BOOK EXAMINATION

This examination accounts for 80% of the module assessment. A total maximum mark obtainable is 100. The marks assigned for each question and parts thereof are indicated in square brackets

If you have any doubt as to the interpretation of the wordings of a question, make your own decision, but clearly state it on the script

Assume reasonable values for any data not given in or provided with the question paper, clearly make such assumptions made in the script

All examinations are conducted under the rules and regulations of the KDU

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Question 1

a. Briefly explain what is meant by the following terms.

i. Digital filter [02]

ii. Recursive filter [02]

iii. Averaging filter [02]

iv. Moving average filter [02]

v. Filter transfer function [02]

b. Consider the following discrete time signal given in Figure 1.

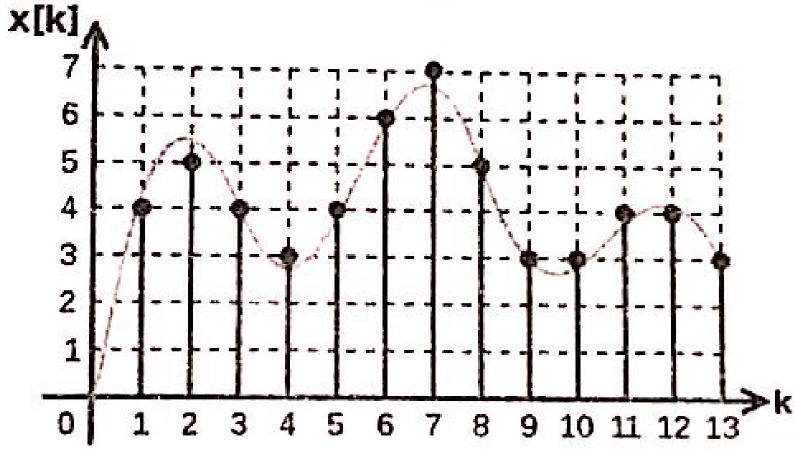

Figure 1.

Calculate the outputs of a 3 point moving average filter at.

i. $k = 7$. [03]

ii. $k = 10$. [03]

c. Draw the ideal filter response of the following frequency selective filters.

i. Bandpass Filter [02]

ii. Notch Filter [02]

d. Answer the following questions considering the electrical circuit given in Figure 2.

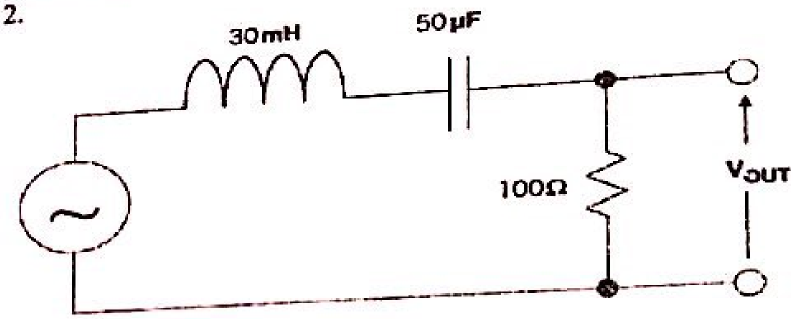

Figure 2.

i. Derive the transfer function of the circuit. [03]

ii. What is the order of this electrical circuit? [02]

Question 2

a. Answer the following questions considering an analog to digital converter.

i. What is meant by analog to digital conversion? [02]

ii. Draw a block diagram of an analog to digital converter indicating the three main blocks of the system. [03]

iii. Briefly explain the functionality of the three main blocks of an analog to digital converter. [03]

b. Consider the signal $x_a(t)$, which is strictly bandlimited so that $X_a(j\Omega) = 0$ for $|\Omega| > \Omega_0$, as indicated in Figure 3.

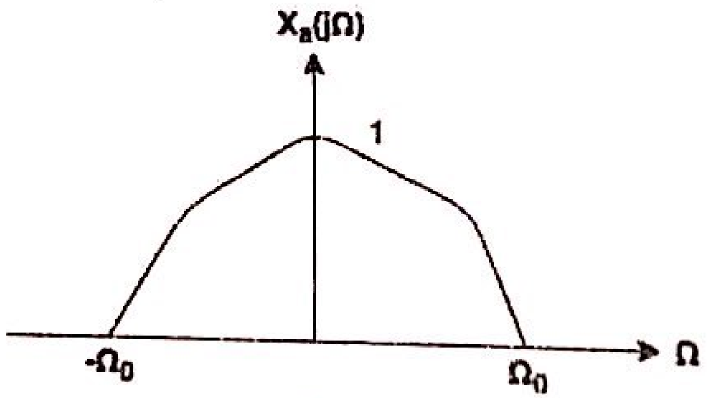

Figure 3.

Draw the Fourier transform of $x_s(T)$ if $x_a(t)$ is sampled with a sampling frequency,

i. $\Omega_s \geq 2\Omega_0$ [03]

ii. $\Omega_s < 2\Omega_0$ [03]

c. Briefly explain what is meant by 'Aliasing' in sampling signals. [02]

d. State the Nyquist sampling theorem. [03]

e. Write down what is meant by 'quantization noise' in analog to digital conversion. [02]

f. Write down the two steps involved in reconstruction process of a signal in a digital to analog converter. [02]

g. What is the use of a zero-order hold for a digital to analog filter? [02]

Question 3

a. Write down the use of Fourier analysis for digital signal processing. [01]

b. Find the inverse discrete Fourier transform function of [04]

$$X(e^{j\omega}) = 5\pi\delta(\omega-\omega_0) + 3\delta(\omega+\omega_0)$$

c. State one advantage of using z-Tranforms compared Fourier transforms, for digital signal processing applications. [02]

d. Find the z-transform of the following sequences.

i. $x(n) = \pi\delta(n) + \delta(n-\pi)$ [04]

ii. $x(n) = \pi^n u(n)$ [04]

e. Briefly explain the following properties of discrete Fourier transform.

i. Linearity. [02]

ii. Symmetry. [02]

iii. Circular shift. [02]

f. Write down the steps of performing linear convolutions using discrete Fourier transform, for the sequence x(n) with N_1 points with system h(n) with N_2 points. [04]

Question 4

a. Consider the linear shift invariant system

$$H(z) = [1 + b(0) + b(1)\, z^{-1}]/[a(0) + a(1)\, z^{-1}]$$

i. Find the linear constant coefficient difference equation relating the input $x(n)$ with the output $y(n)$. [03]

ii. Draw the digital network indicating the system. [04]

iii. Draw the signal flowgraph of the system. [04]

b. Draw the frequency response of a low-pass filter indicating.

i. Passband cutoff frequency ω_p. [02]

ii. Stopband cutoff frequency ω_s. [02]

iii. Passband deviation δ_p. [02]

iv. Stopband deviation δ_s [02]

c. Consider an FIR linear phase low-pass filter having the specifications:

$$0.98 \le |H(e^{j\omega})| \le 1.02,\ 0 \le |\omega| \le 0.28\pi$$

$$|H(e^{j\omega})| \le 0.02,\ 0.32\pi \le |\omega| \le \pi$$

i. Calculate the transition width. [03]

ii. Calculate the cut-off frequency. [03]