Question 2

a. For a sampling rate of 10 kHz, determine the normalized cutoff (in radians/sample) for 1 kHz. [3]

b. State the choice of filter length N you would use for a simple design, and explain your rationale. [3]

c. Write the formula for the ideal impulse response $h_d[n]$ for a low-pass cutoff frequency $\omega_c = \frac{\pi}{10}$ considering both cases when n≠0 and n=0 [4]

$$h_d[n] = \begin{cases} \frac{\sin(\omega_c n)}{\pi n} & n \neq 0 \\ \frac{\omega_c}{\pi} & n = 0 \end{cases}.$$

d. Using a rectangular window, indicate how you would reduce $h_d[n]$ to obtain $h[n]$ [2]
[3]

e. Explain how increasing the filter order N affects the transition bandwidth. [2]

f. Explain how it affects the stop-band attenuation. [3]

g. Comment on the computational cost penalty of a higher order.

Question 3 [4]

a. Find a closed-form expression for the Z-transform $X(z) = \sum_{n=0}^{\infty} \sin(n)\, z^{-n}$ [2]

b. State its region of convergence. [4]

c. For $H(z) = \frac{z+2}{z^2+1.5z+0.5}$, locate its poles and zeros. [3]

d. Sketch the pole-zero plot and use it to assess stability. [4]

e. If unstable, propose one modification to H(z) to ensure stability. [3]

f. Briefly explain why your modification restores stability.

Question 4

a. Compute the linear convolution y[n] = x[n] * h[n] for x[n] = {1, 2, 3}, h[n] = {0, 1, 0.5}. [4]

b. State the length of the result and explain how you determined it. [2]

c. Given impulse response $h[n] = 3^n u(n)$, compute the output y[n] for input x[n] = {1, 1, 1}. [4]

d. Identify whether this Linear Time Invariant (LTI) system is stable. Justify. [3]

e. Define causality for an LTI system in terms of its impulse response. [3]

f. Define Bounded Input Bounded output (BIBO) stability in terms of h[n]. [4]

BINDURA UNIVERSITY OF SCIENCE EDUCATION

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FACULTY OF SCIENCE AND ENGINEERING

DEPARTMENT OF ENGINEERING AND PHYSICS

BACHELOR OF SCIENCE DEGREE IN ELECTRONIC ENGINEERING (HBSCEE)

EEE3103 (2) DIGITAL SIGNAL PROCESSING

DURATION: 3 HOURS TOTAL MARKS: 100

INSTRUCTIONS TO CANDIDATES

1. The paper contains **seven (7)** questions
2. Answer any **five (5)** questions
3. Each question carries **20** marks.

Question 1

a. Define aliasing in discrete-time signals. [2]

b. Give one example of an aliasing artifact in audio or imaging. [2]

c. A signal with maximum frequency 3 kHz is sampled at 6 kHz. Does this satisfy the Nyquist-Shannon theorem? State the condition. [4]

d. Briefly justify your answer in (c). [3]

e. If the sampling rate is reduced to 4 kHz, compute the aliased frequency of the 3 kHz component. [4]

f. Describe qualitatively how this alias would appear in the reconstructed signal. [4]

Question 5

a. For the periodic sequence x[n] = {1, -1, 1, -1} with period N = 4, write its DFS coefficient formula $a_k = \frac{1}{N}\sum x[n]\, e^{-j\left(\frac{2\pi}{N}\right)kn}$. [3]

b. Compute the non-zero DFS coefficients a_k. [4]

c. Compute the 8-point DFT of x[n] = {1, 2, 3, 4, 5, 0, 0, 0}. [4]

d. State the computational complexity difference between direct DFT and FFT for N = 8. [2]

e. Sketch the magnitude spectrum of part d. [3]

f. Explain how the DFT helps identify the signal's frequency components. [4]

Question 6

a. Define an adaptive filter and one typical application. [3]

b. State the cost function minimized by the LMS algorithm, $J = E[e^2[n]]$. [2]

c. Derive the weight-update equation w[n+1] = w[n] + μ e[n] x[n] for LMS. [4]

d. Identify the role of the step-size parameter μ. [2]

e. Outline how LMS is used for noise cancellation in a communications receiver. [4]

f. State one advantage and one drawback of LMS in practical use. [3]

Question 7

a. Explain Quadrature Phase Shift Keying (QPSK) modulation, including symbol mapping. [4]

b. Sketch its constellation diagram with labeled points (+1+j1, etc.). [4]

c. For a Binary Phase Shift Keying (BPSK) system over Additive White Gaussian Noise (AWGN), write the symbol-error rate, $Pe = Q\sqrt{\frac{2E_b}{N_0}}$ [4]

d. Compute the Symbol Error Rate (SER),P_e at Signal to Noise Ratio (SNR) = 10 dB: Pe = Q(√20) ≈ (use table). [3]

e. Describe the function of error-correction codes in digital comms. [3]

f. For the Hamming (7,4) code, what is its code rate and the maximum number of errors it can correct in a single code word? [2]

END OF PAPER