Ref No: Ex/PG/EE/T/1210B/2025

M.E. ELECTRICAL ENGINEERING

FIRST YEAR SECOND SEMESTER EXAMINATION, 2025

ADVANCED DIGITAL SIGNAL PROCESSING

Full Marks 100

Time: Three hours **(50 marks for each part)**

Use a separate Answer-Script for each part

PART I

Answer Question 1 and any one of Questions 2 and 3

1.(a) What conditions must the impulse response sequence and the z-transfer function of a discrete-time linear time-invariant (DTLTI) system satisfy, for the system to be bounded-input bounded output (BIBO) stable? Give relevant derivations.
Explain how the constraint on the transfer function get simplified, for causal systems 15

The difference equation relating the output y[n] and the input x[n] of a DTLTI system is given by

(b) For a DTLTI system the output is

$y[n]=\left[\left(\frac{1}{2}\right)^n-\left(\frac{2}{3}\right)^n\right]u[n]$, when the input is

$$x[n]=\left(\frac{1}{2}\right)^n u[n]+(4)^n u[-n-1]$$

(i) Determine the transfer function of the system.
(ii) Find the impulse response.
(iii) Obtain the difference equation relating the output and the input sequences. 10
(iv) Examine the stability and the causality of the system.

OR

1.(a) Design a low pass digital Butterworth filter with a dc gain of 0 dB, an attenuation of 3 dB at 1620 Hz and an attenuation of at least 10 dB at 3240 Hz. Use impulse invariant transformation. Consider a sampling frequency of 7 kHz. 13
Obtain the difference equation relating the output and the input sequences of the filter.

[Turn over

Ref No: Ex/PG/EE/T/1210B/2025

(b) Examine the correctness of the statement---- A 'digital-to-analog converter' *(DAC) introduces a time-delay of half the sampling period'.*

Explore the type of analog signal processing required after the (DAC) in a digital signal processing system, to minimize the distortions introduced by the converter. Give relevant derivation and sketch in support of your answer. Which of the remedial measures can be best attained by introducing a digital signal processing algorithm prior to the D/A conversion? 4+8

2. (a) "The autocorrelation function $R_X[m]$ of a random process *X[n]*, can be considered as a measure of similarity between the waveforms *X[n]* and *X[n+m]*".-------- Justify or correct the statement with relevant mathematical derivations. 5+5

Show that $|R_X[m]| \leq R_X[0]$.

(b) If X(n) and Y(n) are real jointly wide-sense stationary random processes, determine which of the following expressions are valid, and give reasons. The symbols have their usual meaning.

Give necessary explanations and proofs (if necessary) for this purpose. 10

$$(i) R_X(m) = e^{-|m|}$$
$$(ii) R_{XY}(m) \leq j\sqrt{R_X(0) R_Y(0)}$$
$$(iii) R_X(m) = 2Sin(3m)$$
$$(iv) S_X(e^{j\Omega}) = \frac{6}{6+7\Omega^3}$$
$$(v) S_X(e^{j\Omega}) = 3 + j\Omega^2$$

(c) A DTLTI system is excited by a WSS random process. Derive the expression for the cross-correlation of the output and the input and the cross spectral density, in terms of properties of the input and that of the system. 5
What will this cross-correlation yield if the input is an ideal white noise process?

Ref No: Ex/PG/EE/T/1210B/2025

3. **Write short notes on any TWO of the following.** **12 ½ +12 ½**

(a) **Zero-input limit cycle and dead-band effect in digital IIR filters.**

(b) **Stationarity and ergodicity of random processes.**

(c) **Ideal and bandlimited white noise.**

Ref No: **Ex/PG/EE/T/1210B/2025**

M.E. ELECTRICAL ENGINEERING 1ST YEAR 2ND SEMESTER EXAM 2025

SUBJECT: - ADVANCED DIGITAL SIGNAL PROCESSING

Full Marks 100

Time: Three hours (50 marks for each part)

Use a separate Answer-Script for each part

No. of Questions	PART II	Marks

Answer Q.4 and ANY TWO from the rest.

1. (a) Prove that in a discrete Wiener filter the filter coefficients are obtained by solving the equations: — 09

$$\sum_{m=0}^{N-1} h_m R_{xx}(m-j) = R_{xd}(j), \quad j = 0,1,2,\cdots,(N-1)$$

where each symbol has its usual meaning.

(b) Draw the block diagram of an adaptive noise canceller with correlated noise in the reference input and explain its operating principle. Derive the z-transfer function of this filter when the original signal and the reference input are completely uncorrelated. — 08

2. (a) State and prove how Correlation LMS algorithm can overcome certain limitations of the basic LMS algorithm. In what respect Leaky LMS algorithm bears similarity with RLS algorithm? How are generalized versions of auto-correlation matrix and cross-correlation vector computed to formulate generalized performance criterion in RLS algorithm? — 09

(b) A 5×5 image is given in Fig. 1(a) where $I(k,l)$ gives the pixel intensity at kth row and lth column position. This image is filtered (i) using the mask given in Fig. 1(b) and (ii) using the mask given in Fig. 1(c). Determine the output filtered pixel intensity value at position (2,4) in each case, where original intensity value is 55. — 08

128	78	21	68	45
226	99	116	55	48
54	62	165	75	81
92	41	139	206	177
72	53	234	35	14

Fig. 1(a)

1/16	2/16	1/16
2/16	4/16	2/16
1/16	2/16	1/16

Fig. 1(b)

2/16	2/16	2/16
1/16	4/16	2/16
1/16	1/16	1/16

Fig. 1(c)

Ref No: **Ex/PG/EE/T/1210B/2025**

M.E. ELECTRICAL ENGINEERING 1ST YEAR 2ND SEMESTER EXAM 2025

SUBJECT: - ADVANCED DIGITAL SIGNAL PROCESSING

Full Marks 100

Time: Three hours (50 marks for each part)

Use a separate Answer-Script for each part

No. of Questions	PART II	Marks
3. (a)	How are first derivative image filters implemented using Gradient? How are Sobel operators and Prewitt operators developed based on this principle? Under what circumstances the sum of the coefficients of a high pass spatial image filter mask will give 0 and under what circumstances it will give 1?	08
(b)	In image processing what are edge pixels, edges and edge detectors? How can basic Global Processing be employed for edge linking and boundary detection and what is its limitation? Give a detailed description of how Hough Transform can be employed for the same purpose to overcome this limitation.	09
4.	Write short notes on ***any two*** of the following: *i*) Necessary condition for convergence in a steepest descent based adaptive digital filter. *ii*) Direct realization of linear-phase FIR digital filters. *iii*) Fourier series for a periodic discrete sequence.	08×2 =16