Hall Ticket No | | | | | | | | | | | Question Paper Code: AEC012

INSTITUTE OF AERONAUTICAL ENGINEERING

(Autonomous)

Dundigal, Hyderabad - 500 043

MODEL QUESTION PAPER

B.Tech VI Semester End Examinations, April - 2020

Regulation: IARE-R16

DIGITAL SIGNAL PROCESSING

(Electronics and Communication Engineering)

Time: 3 Hours **Max Marks: 70**

Answer any ONE question from each Unit

All questions carry equal marks

All parts of the question must be answered in one place only

UNIT – I

1 a) Determine the impulse response and step response of the causal system given below and discuss on stability: [7M]

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

b) The impulse response of LTI system is h(n)={1 2 1-1} Determine the response of the system if input is x(n)={1 2 3 1} [7M]

2 a) Determine the output y(n) of LTI system with impulse response h(n)= an u(n). $|a|<1$ When the input is unit input sequence that is x(n)=u(n) [7M]

b) Obtain the Direct form II of y (n) = -0.1(n-1) + 0.72 y(n-2) + 0.7x(n) -0.252 x(n-2) [7M]

UNIT – II

3 a) Develop a 8 point DIT-FFT algorithm. Draw the signal flow graph. [7M]

b) Given two sequences $x_1(n)$and $x_2(n)$ of length N obtain expression to compute circular convolution these sequences. What are the changes required if circular convolution output same as linear convolution output. [7M]

4 a) Find DFT of following sequence $x(n)$ for N=4 and N=8 and plot magnitude of DFT X(k) and comments on results obtained. $x(n) = \begin{cases} 1 & \text{for } 0 \le n \le 2 \\ 0 & \text{for other wise} \end{cases}$ [7M]

b) Compute the IDFT using DIF FFT algorithm given that X (k) = { 4, 1-j2.414, 0, 1-j0.410, 0, 1+j0.414,0, 1+j2.414}. [7M]

UNIT – III

5 a) Design an analog Butterwoth filter has a -2db passband attenuation at a frequency of 20 rad/sec. and at least -10db stop band attenuation at 30 rad/sec. [7M]

b) Discuss & Explain Transformation of Analog filters into equivalent digital filters using Bilinear transformation method. [7M]

6 a) Design a 3rd order Butterworth digital filter using Impulse Invariant technique , assume sampling T is 1 sec. and for N is 3. [7M]

b) Discuss & Explain Transformation of Analog filters into equivalent digital filters using impulse invariant method. [7M]

UNIT – IV

7 a) Design an ideal high pass filter with a frequency respose Hd(ejw)=1 for $\frac{\pi}{4} \leq |\omega| \leq \pi$, 0 for $|\omega| \leq \frac{\pi}{4}$. Find the values of h(n) for N=11.Find H(z).plot magnitude response.. [7M]

b) Using a rectangular window technique design a low pass filter with pass band gain of unity, cutoff frequency of 100Hz and working at a sampling frequency of 5KHz.The length of the impulse response should be 7. [7M]

8 a) Design a HPF of length 7 with cut off frequency of 2 rad/sec using Hamming window. Plot the magnitude and phase response. [7M]

b) Design a high pass filter using hamming window with a cut-off frequency of 1.2radians/second and N=9 [7M]

UNIT – V

9 a) With the help of block diagram explain the sampling rate conversion by a rational factor 'I/D'. Obtain necessary expressions. [7M]

b) A digital system is characterized by the difference equation y(n)=0.95y(n-1)+x(n).determine the dead band of the system when x(n)=0 and y(-1)=13. [7M]

10 a) Write short note on (i) Truncation and rounding (ii) Coefficient Quantization. [7M]

b) Describe the decimation process with a neat block diagram. Consider a signal x(n)= sin(πn)U(n). Obtain a signal with an interpolation factor of '2' [7M]

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COURSE OBJECTIVES:

The course should enable the students to:

I	Provide background and fundamental material for the analysis and processing of digital signals and to familiarize the relationships between continuous-time and discrete-time signals and systems.
II	Study fundamentals of time, frequency and z-plane analysis and to discuss the inter-relationships of these analytic method and to study the designs and structures of digital (IIR and FIR) filters from analysis to synthesis for a given specifications.
III	Introduce a few real-world signal processing applications.
IV	Acquaint in FFT algorithm, multi-rate signal processing techniques and finite word length effects.

COURSE OUTCOMES (COs):

CO1	Interpret, represent and process discrete/digital signals and systems
CO2	Understanding of time domain and frequency domain analysis of discrete time signals and systems.
CO3	Understand DFT for the analysis of digital signals & systems
CO4	Demonstrate and analyze DSP systems like FIR and IIR Filter
CO5	Understand multi rate signal processing of signals through systems.

COURSE LEARNING OUTCOMES:

AEC012.01	Understand how digital to analog (D/A) and analog to digital (A/D) converters operate on a signal and be able to model these operations mathematically.
AEC012.02	Define simple non-periodic discrete-time sequences such as the impulse and unit step, and perform time shifting and time-reversal operations on such sequences.
AEC012.03	Given the difference equation of a discrete-time system to demonstrate linearity, time-invariance, causality and stability, and hence show whether or not a given system belongs to the important class of causal, LTI (linear time-invariant) systems.
AEC012.04	Given the impulse response of a causal LTI system, show whether or not the system is bounded-input/bounded-output (BIBO) stable.
AEC012.05	Perform time, frequency and Z-transform analysis on signals.
AEC012.06	From a linear difference equation of a causal LTI system, draw the Direct Form I and Direct Form II filter realizations.
AEC012.07	Knowing the poles and zeros of a transfer function, make a rough sketch of the gain response.
AEC012.08	Define the Discrete Fourier Transform (DFT) and the inverse DFT (IDFT) of length N.
AEC012.09	Understand the inter-relationship between DFT and various transforms.
AEC012.10	Understand the significance of various filter structures and effects of round-off errors.
AEC012.11	Understand the fast computation of DFT and appreciate the FFT Processing.
AEC012.12	Design of infinite impulse response (IIR) filters for a given specification.
AEC012.13	Design of finite impulse response (FIR) filters for a given specification.
AEC012.14	Compare the characteristics of IIR and FIR filters.
AEC012.15	Understand the tradeoffs between normal and multi rate DSP techniques and finite length word effects.

AEC012.16	Understand the signal interpolation and decimation, and explain their operation
AEC012.17	Explain the cause of limit cycles in the implementation of IIR filters.

MAPPING OF SEMESTER END EXAMINATION TO COURSE LEARNING OUTCOMES:

SEE Question No.		Course Learning Outcomes		COs	Blooms Taxonomy Level
1	a	AEC012.02	Define simple non-periodic discrete-time sequences such as the impulse and unit step, and perform time shifting and time-reversal operations on such sequences.	CO 1	Understand
	b	AEC012.03	Given the difference equation of a discrete-time system to demonstrate linearity, time-invariance, causality and stability, and hence show whether or not a given system belongs to the important class of causal, LTI (linear time-invariant) systems.	CO 1	Understand
2	a	AEC012.03	Given the difference equation of a discrete-time system to demonstrate linearity, time-invariance, causality and stability, and hence show whether or not a given system belongs to the important class of causal, LTI (linear time-invariant) systems.	CO 1	Understand
	b	AEC012.06	From a linear difference equation of a causal LTI system, draw the Direct Form I and Direct Form II filter realizations.	CO 1	Understand
3	a	AEC012.11	Understand the fast computation of DFT and appreciate the FFT Processing.	CO 2	Understand
	b	AEC012.11	Understand the fast computation of DFT and appreciate the FFT Processing.	CO 2	Understand
4	a	AEC012.11	Understand the fast computation of DFT and appreciate the FFT Processing.	CO 2	Understand
	b	AEC012.11	Understand the fast computation of DFT and appreciate the FFT Processing.	CO 2	Remember
5	a	AEC012.13	Design of finite impulse response (FIR) filters for a given specification.	CO 3	Understand
	b	AEC012.14	Compare the characteristics of IIR and FIR filters.	CO 3	Remember
6	a	AEC012.13	Design of finite impulse response (FIR) filters for a given specification.	CO 3	Understand
	b	AEC012.14	Compare the characteristics of IIR and FIR filters.	CO 3	Understand
7	a	AEC012.13	Design of finite impulse response (FIR) filters for a given specification.	CO 4	Understand
	b	AEC012.13	Design of finite impulse response (FIR) filters for a given specification.	CO 4	Understans
8	a	AEC012.13	Design of finite impulse response (FIR) filters for a given specification.	CO 4	Understand
	b	AEC012.13	Design of finite impulse response (FIR) filters for a given specification.	CO 4	Understand
9	a	AEC012.17	Explain the cause of limit cycles in the implementation of IIR filters.	CO 5	Understand
	b	AEC012.16	Understand the signal interpolation and decimation, and explain their operation	CO 5	Remember
10	a	AEC012.14	Compare the characteristics of IIR and FIR filters.	CO 5	Understand
	b	AEC012.14	Compare the characteristics of IIR and FIR filters.	CO 5	Remember

Signature of Course Coordinator **HOD, ECE**