Q.P. Code: 16EC422 **R16**

Reg. No: | | | | | | | | | | |

SIDDHARTH INSTITUTE OF ENGINEERING & TECHNOLOGY:: PUTTUR

(AUTONOMOUS)

B.Tech III Year II Semester Regular & Supplementary Examinations October-2020

DIGITAL SIGNAL PROCESSING

(Electronics and Communication Engineering)

Time: 3 hours Max. Marks: 60

(Answer all Five Units **5 x 12 = 60** Marks)

UNIT-I

1	**a**	Determine the linear convolution for the two sequences x(n)={3,2,1,2},h(n)= {1,2,1,2}.	**7M**
	b	Explain the classification of discrete-time signals.	**5M**

OR

2	**a**	Find 8 point DFT of the sequence x(n)=[1,2,1,0,2,3,0,1].	**7M**
	b	Describe the relation between i) DFT to Z- transform ii) DFT to Fourier Series.	**5M**

UNIT-II

3	**a**	Describe the relation between i) DFT to Z- transform ii) DFT to Fourier Series.	**7M**
	b	Compare DFT and FFT algorithms.	**5M**

OR

4	**a**	Explain Radix-4 FFT algorithm in decimation in time domain.	**6M**
	b	Compare radix-2 DIT-FFT and DIF-FFT algorithms.	**6M**

UNIT-III

5 Obtain direct form-I, direct form-II, cascade, parallel form realization of following system: $y(n) = 0.75y(n-1)-0.125y(n-2)+3x(n)+7x(n-1)+x(n-2)$ **12M**

OR

6 Realize system with following difference equation **12M**
y(n) = (3/4) y(n-1) – (1/8) y(n-2) + x(n) + (1/3) x(n-1).
a) Cascade form
b) Parallel form

UNIT-IV

7	**a**	Determine the order and the pole of the low pass filter that has a 3-dB attenuation at 500 Hz and an attenuation of 40 dB at 1000 Hz.	**7M**
	b	Distinguish the Butterworth and Chebyshev filters.	**5M**

OR

8	**a**	The analog transfer function H(s)= 2/(s+1)(s+2) Determine H(z) using impulse invariance method.	**7M**
	b	Compare FIR and IIR filters.	**5M**

UNIT-V

9	**a**	Discuss about characteristics linear phase FIR filters.	**6M**
	b	Design a linear phase FIR filter using frequency sampling method.	**6M**

OR

10 A band pass FIR filter of length 7 is required. It is to have lower and upper cut off frequencies of 3kHz and is intended to be used with a sampling frequency of 24kHz. Determine the filter coefficients using hamming window. Consider the filter to be causal. **12M**

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