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Total No. of Questions – [08] G.R. No.

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Paper Codet U359-131 (ESE)

## **DECEMBER 2019/ENDSEM**

T. Y. B. TECH. (E & TC) (SEMESTER - I) COURSE NAME: Discrete Time Signal Processing COURSE CODE: ETUA31171

## (PATTERN 2017)

Time: [2 Hours]

[Max. Marks: 50]

- (\*) Instructions to candidates:
  1) Answer 0.1, 0.2, 0.3, 0.4, 0.5 OR 0.1
- Answer Q.1, Q.2, Q.3, Q.4, Q.5 OR Q.6, Q.7 OR Q.8
   Figures to the right indicate full marks
- Figures to the right indicate full marks.
   Use of scientific calculator is allowed
- 4) Use suitable data where ever required
- Q.1) a) The analog signal is represented as

[6 marks]

[6 marks]

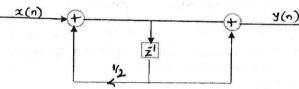
i. What is the Nyquist rate of the signal

 $x(t) = \sin(100\pi t) + 2\sin(20\pi t) - 2\cos(30\pi t)$ 

- ii. If the signal is sampled with sampling frequency 20Hz, what is the discrete time signal obtained after sampling?
- iii. What is the recovered signal?

## OR

b) Consider the discrete time system shown below:



- i) Determine the input output relation.
- ii) Compute first four samples of impulse response
- iii) Apply input x(n) = u(n) and calculate first four samples of output
- Q.2) a) Using DFT and IDFT, calculate circular convolution of sequences  $x1(n) = \{1,1,2,2\}$  and  $x2(n)=\{1,2,3,4\}$  [6 marks]
  - b) Compute 4 p mt DFT of x(n)=cos $\left(\frac{\pi}{2}\right)n$ , using DIT FFT. [6 marks] Compare the computations with Direct DFT approach
- Q.3) a) Compute the response of the system [6 marks] y(n)= 0.7y(n-1)- 0.12y(n-2) + x(n-1) + x(n-2) to an input nu(n). Is the system stable?

OR

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- b) Determine the signal whose Z transform is given by X(Z) = log(1+az<sup>-1</sup>) Hint: Use Differentiation property [6 marks]
- Q.4) a) An analog filter has a transfer function  $(s) = \frac{1}{s+1}$ . [4 marks]

Using bilinear transformation determine digital transfer function of the filter and also write difference equation of the filter. Assume T=1.

OR

b) Draw cascade realization of H(Z)=  $\frac{3z^2+3.6z-0.6}{z^2+0.1z-0.2}$ 

[4 marks]

- Q. 5) a) Design a digital high pass filter to meet the following specifications. Cut off frequency = 250Hz, Sampling rate 1000smaples/sec, N =7. Use Hamming window.
  - b) Transfer function of FIR filter has 2 poles at z=0 and two zeros [4 marks] at z = -1 with DC gain of 8. Find the transfer function and impulse response of the filter. Is it a causal filter? Is it linear phase filter?
  - c)Explain Gibb's phenomenon in windowing technique of [4 marks] FIR filters

OR

- Q.6) a) Using frequency sampling method, design a band pass filter [6 marks] With, Fs = 8000Hz, Fc1 = 1000Hz, Fc2 = 3000Hz, N=7.
  - b) A linear phase FIR filter rejects a frequency component at  $\omega_0 = \frac{2\pi}{3}$ . [4 marks] It's frequency response is normalized so that H(0)=1.Find the transfer Function and impulse response of the filter.
  - c) Compare FIR and IIR filters

[4 marks]

[4 marks]

[4 marks]

Q.7) a) Sampling rate of a signal is to be reduced from 96 KHz to 1KHz. [6 marks] Frequency band of interest is up to 450Hz.  $\delta p = 0.01$  and  $\delta s = 0.001$ . Design a two stage decimator with decimation factors 32 and 3 respectively for stage1 and stage2.

b) Explain sampling rate conversion by non-integer factor

- c) Justify the statement mathematically: Decimation process is time variant.
  - OR
- Q.8) a) Draw the block diagram of decimation process and draw the [6 marks] Spectra of signals at each stage.
  - b) Sampling rate of a signal is to be reduced from 96 KHz to 1KHz. [4 marks] Frequency band of interest is up to 450Hz.  $\delta p = 0.01$  and  $\delta s = 0.001$ . Design a single stage decimator
  - c) Discuss, how multirate signal processing is applied in [4 marks]
     CD Hifi systems.

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