



Regulation R17

TKR COLLEGE OF ENGINEERING AND TECHNOLOGY
(Autonomous, Accredited by NAAC with 'A' Grade)

Subject code: 1E5BE

B.Tech III Year I Semester Supplementary Examinations, December 2021

DIGITAL SIGNAL PROCESSING

(Electrical and Electronics Engineering)

Maximum Marks: 70

Date: 08.02.2022 Duration: 3 hours

- Note:
1. This question paper contains two parts A and B.
 2. Part A is compulsory which carries 20 marks. Answer all questions in Part A.
 3. Part B consists of 5 Units. Answer any one full question from each unit.
 4. Each question carries 10 marks and may have a, b, c, d as sub questions.

Part-A

All the following questions carry equal marks

(10x2M=20 Marks)

- 1 Differentiate between Analog and Discrete signal.
- 2 Define the Stability in the system.
- 3 Why discrete Fourier series are needed?
- 4 Differentiate between Linear and Circular convolution?
- 5 What is meant by bilinear transformation technique?
- 6 Distinguish between Analog filters and Digital filters.
- 7 What are the disadvantages of Fourier series method?
- 8 What are the desirable characteristics of the window?
- 9 What is the need for anti-imaging filter after up sampling a signal?
- 10 What are the effects of Dead band?

Part-B

Answer All the following questions.

(5X10M=50Marks)

- 11 a) Explain the block diagram of DSP and its applications. (5M)
b) Define & Test for Linearity, Causality & Time variance and Invariance for the given systems described by: (5M)

i) $y(n) = n^2 x(n)$ ii) $y(n) = x(n-1)$

OR

- 12 Find the magnitude and phase response of the system for given equation. (10M)

$$y(n) = 1/2 [x(n) + x(n-2)]$$

- 13 Find the Circular convolution between given sequences using any one of the methods and compare with Linear Convolution output: (10M)

$$x(n) = \{1, 0, -2, -1, 0\}, h(n) = \{-1, 0, 1, 1\}.$$

OR

- 14 Find the DFT of a given signal $x(n) = \{1, 0, 1, 2, 0, 0, 1, 0\}$ by using DIT-FFT algorithm. (10M)

- 15 a) What are the advantages and disadvantages of digital filters. (4M)
b) Design Butterworth low pass filter with the given specifications $\alpha_p = 0.5\text{dB}$, $\alpha_s = 20\text{dB}$ and $\Omega_p = 150\text{rad/sec}$, $\Omega_s = 300\text{rad/sec}$. Convert this filter into digital filter by using step invariant method. (6M)

OR

- 16 Realize the following system (10M)
 $y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.252x(n-2)$ by using
i) Direct form-I and ii) Direct form-II
- 17 a) Explain the characteristics of FIR filter. (4M)
b) Design an ideal low pass filter whose frequency response

$$H_d(e^{j\omega}) = 1 \text{ for } -\pi/2 \leq \omega \leq \pi/2$$
$$= 0 \text{ for } \pi/2 \leq \omega \leq \pi.$$

Find the values of $h(n)$ for $N=11$. Find $H(Z)$ Using Hamming window. (6M)

OR

- 18 a) Explain the Design of FIR digital filter by using fourier series method. (5M)
b) What do you understand by linear phase response. (5M)
- 19 What is the need of Multirate signal processing? Explain the advantages of lower sample rates and higher sample rates (10M)

OR

- 20 a) Explain limit cycles. (5M)
b) Explain what are the methods used to prevent overflow. (5M)