Code No: 156AR **R18**

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD
B. Tech III Year II Semester Examinations, January/February - 2025
DIGITAL SIGNAL PROCESSING
(Common to ECE, EIE)

Time: 3 Hours **Max. Marks: 75**

Note: i) Question paper consists of Part A, Part B.
ii) Part A is compulsory, which carries 25 marks. In Part A, Answer all questions.
iii) In Part B, Answer any one question from each unit. Each question carries 10 marks and may have a, b as sub questions.

PART – A

(25 Marks)

1.a) Define sampling and state the Nyquist sampling theorem. [2]
b) State the convolution property of the Z-transform. [3]
c) Define Discrete Fourier Series. [2]
d) What are the basic properties of the Discrete Fourier Transform (DFT)? [3]
e) What is the main advantage of using the Bilinear Transformation method in IIR filter design? [2]
f) Define the Butterworth filter. What is its key characteristic? [3]
g) Define FIR digital filters. How do they differ from IIR filters? [2]
h) What is a Kaiser window? In what way is it superior to another window function? [3]
i) Define the stability criterion of a digital filter. How is it determined? [2]
j) How do round-off errors affect the performance of IIR filters? [3]

PART – B

(50 Marks)

2. Find the impulse response and step response of a discrete-time LTI system whose difference equation is given by $y(n) = y(n-1) + 0.5y(n-2) + x(n) + x(n-1)$. [10]

OR

3.a) Find the z-transform of $x(n) = 2^n$ for $n \geq 0$, and calculate its Region Of Convergence (ROC).
b) Discuss the properties of the unit step function in discrete time. [6+4]

4.a) Use the Overlap-Add method to compute the DFT of the sequence $x(n)$= {1,1,0,0,1,1,0,0} with a block length of 4.
b) Discuss the importance of the Fast Fourier Transform (FFT) in digital signal processing. [6+4]

OR

5. Given the sequence $x(n)$= {1,2,3,4,5}, calculate the DFT using the Radix-2 Decimation-in-Time (DIT) FFT algorithm. Show each step of the computation. [10]

6.a) Explain the characteristics of a Chebyshev filter and how it differs from a Butterworth filter.
b) For the Analog Transfer function $H(s) = 2\frac{(s+1)}{(s+2)}$ determine $H(z)$ using Impulse Invariance method. Assume T=1sec. [4+6]

OR

7.a) Design a Butterworth Low-pass Filter, given specifications α_p=3dB, α_s=18dB, f_p=2KHz, f_s=3KHz. Find the order of butter worth filter.

b) What is meant by frequency warping? What is the cause of this effect? [6+4]

8. Design a high-pass FIR filter using the frequency sampling technique. The filter should have a cut-off frequency of 2 kHz, and the sampling rate is 10 kHz. Explain all the design steps involved. [10]

OR

9.a) Explain the procedure for designing an FIR filter using the Fourier method.

b) Discuss in detail the different types of window functions used in FIR filter design. Compare the effects of each window on the filter performance. [4+6]

10.a) Explain the limit cycle oscillations due to product round off and overflow errors.

b) Solve the following difference equation using Z-transform:

$$y(n) - 1.5y(n-1) + 0.75y(n-2) = x(n)$$

Find the system function H(z) and its frequency response. [4+6]

OR

11.a) Describe the direct form and canonic form of digital filter realization and compare them.

b) Derive the realization of a digital filter in Direct Form-II using the system function $H(z) = \frac{(z-0.5)}{(z+1)}$. [4+6]

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