Subject Code: BEC503

PAPER ID-311394

Roll No:

BTECH
(SEM V) THEORY EXAMINATION 2025-26
DIGITAL SIGNAL PROCESSING

TIME: 3 HRS **M.MARKS: 70**

Note: Attempt all Sections. In case of any missing data; choose suitably.

SECTION A

1. Attempt *all* questions in brief. 2 x 07 = 14

Q no.	Question	CO	Level
a.	Write down the real time applications of digital signal processing.	1	K1
b.	Why realization process is used in digital system	1	K1
c.	Differentiate between filter and frequency transformation.	2	K1
d.	What are the advantages and disadvantages of IIM.	2	K1
e.	What is Gibbs phenomenon with diagram?	3	K1
f.	What is circular convolution?	4	K1
g.	What is the need of multirate signal processing?	5	K1

SECTION B

2. Attempt any *three* of the following: 07 x 3 = 21

a.	Explain the linear phase property of FIR filter. Also determine linear-phase and cascade realizations of the system function given below. $H(z) = 1 + \frac{3}{4}z^{-1} + \frac{17}{8}z^{-2} + \frac{3}{4}z^{-3} + z^{-4}$	1	K3
b.	Convert a low pass Butterworth filter into a BPF with upper and lower cut off frequencies are 0.6π and 0.4π respectively. The LPF has 3 dB bandwidth =0.2 π. $H(z) = \frac{0.245(1+z^{-1})}{(1-0.509z^{-1})}$	2	K3
c.	Explain the characteristics of kaiser window. Briefly explain the design steps for designing of finite impulse response filter using kaiser window with suitable diagram.	3	K2
d.	Derive the equation for the DIT algorithm for N = 8 and draw the signal flow graph.	4	K3
e.	Consider time domain sequence x(n)= {1,3,5,7,9}. Determine the up sampled version of the signal for sampling rate multiplication factor a) I=2 (b) I=3 (c) I=4.	5	K2

SECTION C

3. Attempt any *one* part of the following: 07 x 1 = 07

a.	Obtain the direct form-I , direct form-II, cascade form realization structures for the following system. $y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.25x(n-2)$	1	K3

Subject Code: BEC503

PAPER ID-311394

Roll No:

BTECH
(SEM V) THEORY EXAMINATION 2025-26
DIGITAL SIGNAL PROCESSING

TIME: 3 HRS **M.MARKS: 70**

b.	Determine the coefficients of a continued-fraction expansion of H(z); Also draw ladder realization structure of a given IIR system. $$H(z) = \frac{2+8z^{-1}+6z^{-2}}{(1+8z^{-1}+12z^{-2})}$$	1	K3

4. Attempt any *one* part of the following: 07 x 1 = 07

a.	Design a digital Chebyshev filter to satisfy the constraints using bilinear transformation with T=1 sec $$0.707 \le \left|H(e^{j\omega})\right| \le 1 \qquad 0 \le \omega \le 0.2\pi$$ $$\left|H(e^{j\omega})\right| \le 0.1, \qquad 0.5\pi \le \omega \le \pi$$	2	K3
b.	Find the system function H(z) of the digital Butterworth filter that meets the following specification: Use Bilinear transformation method. Assume T=1 sec a) 1 dB ripple in pass band $0 \le \lvert\omega\rvert \ge 0.3\pi$ b) At least 40 dB attenuation in stop band $0.8\pi \le \lvert\omega\rvert \ge \pi$,	2	K2

5. Attempt any *one* part of the following: 07 x 1 = 07

a.	Explain Butterworth and Chebyshev IIR filter design.	3	K2
b.	Design a linear phase low pass digital filter if the desired frequency response is giving by $$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} & 0 \le \lvert\omega\rvert \le \frac{\pi}{2} \\ 0 & \frac{\pi}{2} < \lvert\omega\rvert \le \pi \end{cases}$$ Using the bartlett window and choosing a suitable length of filter length M, find the impulse response and frequency response of designed filter. Determine the system function and difference equation. Also draw the linear phase structure of designed filter.	3	K3

6. Attempt any *one* part of the following: 07 x 1 = 07

a.	Compute IDFT of the sequence X(k)= {7, -0.707-j0.707, -j, 0.707-j0.707, 1, 0.707+j0.707, j, -0.707+j0.707}, using FFT Algorithm.	4	K3
b.	An input sequence x(n) = {2, 1, 0, 1, 2} is applied to a DSP system having an impulse sequence h(n) = {5, 3, 2, 1}. Determine the output sequence by linear convolution and verify the same through circular convolution.	4	K3

7. Attempt any *one* part of the following: 07 x 1 = 07

a.	Explain the concept of Quadrature Mirror Filters (QMF) used in multirate digital signal processing.	5	K3
b.	Briefly explain the phenomenon of Subband coding of speech signals with neat diagram. Also explain the application of Subband coding of speech signal in multirate signal processing.	5	K2