



PAPER ID-311394

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Subject Code: BEC503

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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 $= 0.2\pi$. $H(z) = \frac{0.245(1+z^{-1})}{(1-0.509z^{-1})}$	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
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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
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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 $0.707 \leq H(e^{j\omega}) \leq 1 \quad 0 \leq \omega \leq 0.2\pi$ $ H(e^{j\omega}) \leq 0.1, \quad 0.5\pi \leq \omega \leq \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 $0 \leq \omega \leq 0.3\pi$ b) At least 40 dB attenuation in stop band $0.6\pi \leq \omega \leq \pi$	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 $H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} & 0 \leq \omega \leq \frac{\pi}{2} \\ 0 & \frac{\pi}{2} < \omega \leq \pi \end{cases}$ 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 $X(k) = \{1, -0.707-j0.707, -j, 0.707-j0.707, 1, 0.707+j0.707, j, -0.707+j0.707\}$, 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