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BTECH
(SEM V) THEORY EXAMINATION 2023-24
DIGITAL SIGNAL PROCESSING

TIME: 3 HRS**M.MARKS: 100**

Note: 1. Attempt all Sections. If require any missing data; then choose suitably.

SECTION A**1. Attempt all questions in brief.****2 x 10 = 20**

Q no.	Question	Marks	CO
a.	Distinguish between IIR and FIR filter.	2	1
b.	Define linear phase response of a filter.	2	1
c.	Compare bilinear transformation and Impulse invariant method of IIR filter design.	2	2
d.	Distinguish between Butterworth and Chebyshev (Type-I) filter.	2	2
e.	State the need for employing window for designing FIR filter?	2	3
f.	The most straight forward approach to filter design is to truncate the impulse response of an ideal IIR filter. Why this is usually an undesirable approach?	2	3
g.	Develop the 4-point DFT of the sequence $x(n) = \{1, 1\}$.	2	4
h.	How is the FFT algorithm applied to determine inverse discrete Fourier transform?	2	4
i.	Highlight the features of a commercial digital signal processor.	2	5
j.	Explain the concept of multistage sampling rate conversion.	2	5

SECTION B**2. Attempt any three of the following:****10x3=30**

a.	Determine the cascade and parallel realization for the system transfer function $H(z) = \frac{3(2z^2 + 5z + 4)}{(2z + 1)(z + 2)}$	10	1
b.	Use bilinear transformation to convert low pass filter $H(s) = \frac{1}{(1 + 1.41s + s^2)}$ into a high pass filter with pass band edge at 100 Hz and $F_s = 1$ kHz.	10	2
c.	Design a low pass FIR filter for the following specifications using rectangular window function. Cut-off frequency = 500 Hz; Sampling frequency = 2000 Hz; Order of the filter = 10	10	3
d.	The first five points of the 8-point DFT of a real valued sequence are $\{0.25, 0.125 - j0.3018, 0, 0.12 - j0.0518, 0\}$. Determine the remaining three points.	10	4
e.	Explain the various types of addressing modes of digital signal processor with suitable example.	10	5

SECTION C**3. Attempt any one part of the following:****10x1=10**

a.	Realize the given $H(z)$ for using ladder structure. $H(z) = \frac{2 + 8z^{-1} + 6z^{-2}}{1 + 8z^{-1} + 12z^{-2}}$	10	1
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b.	Obtain the direct form-I and direct form-II realization of a given LTI system: $y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.25x(n-2)$	10	1
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4. Attempt any one part of the following:**10x1=10**

a.	Convert the analog filter with system function $H_a(s) = \frac{(s + 0.1)}{(s + 0.1)^2 + 9}$ into a digital IIR filter by means of the impulsive invariance method.	10	2
b.	Design a Chebyshev filter for the following specification using bilinear transformation. $0.8 \leq H e^{j\omega} \leq 1 \quad 0 \leq \omega \leq 0.2\pi$ $ H e^{j\omega} \leq 0.2 \quad 0.6\pi \leq \omega \leq \pi$	10	2

5. Attempt any one part of the following:**10x1=10**

a.	What is Hamming Window Function? Obtain its frequency domain characteristics.	10	3
b.	Design a low pass filter using Kaiser window satisfying the specifications given filter: Passband cutoff frequency $F_p = 150$ Hz Stopband cutoff frequency $F_s = 250$ Hz Sampling frequency $F_t = 1000$ Hz Passband attenuation $A_p = 0.1$ dB Stopband attenuation $A_s = 40$ dB	10	3

6. Attempt any one part of the following:**10x1=10**

a.	Determine the DFT of the given sequence $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ using DIF FFT algorithm.	10	4
b.	Find the output $y(n)$ of a filter whose impulse response is $h(n) = \{1, 1, 1\}$ and input signal $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ using overlap add method.	10	4

7. Attempt any one part of the following:**10x1=10**

a.	Explain the process of multirate signal processing in detail. Also, enlist the advantages of multirate signal processing.	10	5
b.	Write the short note on: (i) Recursive Least Square Algorithm (ii) Window LMS Algorithm	10	5