Subject Code: KEC503

PAPER ID-310681

Roll No:

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 algorithm applied to determine inverse discreate 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 $$\mathrm{H(z)} = \frac{3(2z^2+5z+4)}{(2z+1)(z+2)}$$	10	1
b.	Use bilinear transformation to convert low pass filter $$\mathrm{H(s)} = \frac{1}{(1+1.41s+s^2)}$$ into a high pass filter with pass band edge at 100 Hz and Fs=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. $$\mathrm{H(z)} = \frac{2+8z^{-1}+6z^{-2}}{1+8z^{-1}+12z^{-2}}$$	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

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. $0.8 \le \lvert He^{jw}\rvert \le 1$ $\quad 0 \le w \le 0.2\pi$ $\lvert He^{jw}\rvert \le 0.2$ $\quad 0.6\pi \le w \le \pi$	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 = 0.1dB 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) (ii) Window LMS Algorithm	10	5