Subject Code: KEC503

PAPER ID-310692

Roll No:

B. TECH
(SEM-V) THEORY EXAMINATION 2020-21
DIGITAL SIGNAL PROCESSING

Time: 3 Hours | ***Total 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.	What are the advantages and disadvantages of digital signal processing?	2	1
b.	Distinguish between recursive and non-recursive structure used for the realization of digital system.	2	1
c.	What are the differences between impulse invariant transformation and bilinear transformation method?	2	2
d.	Explain the phenomenon of digital frequency transformation.	2	2
e.	What is Gibb's phenomenon in FIR filters?	2	3
f.	What is the dead band effect in digital filters?	2	3
g.	Explain the terms: (i)Ccomputations in one place, (ii) Bit reversal.	2	4
h.	Compute the 4-point DFT of the Following sequence x(n)=cos(nπ) using linear transformation matrix.	2	4
i.	Explain the concept of multistage sampling rate conversion.	2	5
j.	Enlist the various features of digital signal processor.	2	5

SECTION B

2. Attempt any *three* of the following: 3 x 10 = 30

Q no.	Question	Marks	CO
a.	Determine the coefficients of a continued-fraction expansion of H (z); Also draw ladder realization structure of H(z). $$H(z)=\frac{2+8z^{-1}+6z^{-2}}{(1+8z^{-1}+12z^{-2})}$$	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 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	10	3
d.	What are the advantages of FFT over DFT? Explain DIT. Derive the equation for DIT algorithm for N = 8 and draw the signal flow graph.	10	4
e.	Explain the process of multirate signal processing in detail. Also enlist the advantages of multirate signal processing.	10	5

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SECTION C

3. Attempt any *one* part of the following:

Q no.	Question	Marks	CO
a.	Obtain the direct form-I, direct form-II, cascade, and parallel form 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
b.	For $H(z) = 1 + 2z^{-1} - z^{-2} + 3z^{-3} + 3z^{-4} - z^{-5} + 2z^{-6} + z^{-7}$ Draw the direct form and linear form FIR implementation. Also compare the implementation.	10	1

4. Attempt any *one* part of the following:

a.	Compute the poles of an analog Chebyshev filter transfer function that satisfies the constraints: **Passband:** $0.8 \le \lvert H(e^{j\omega})\rvert \le 1 \quad \lvert\omega\rvert \le 0.2\pi$ **Stopband:** $\lvert H(e^{j\omega})\rvert \le 0.2 \quad 0.32\pi \le \lvert\omega\rvert \le \pi$ And determine H(s) and hence obtain H(z) using Bilinear transformation. Assume T=1 sec.	10	2
b.	Design a digital low pass Butterworth IIR filter using impulse invariant method for the following specification. (assume T=1 sec) **Passband:** $0.8 \le \lvert H(e^{j\omega})\rvert \le 1 \quad \lvert\omega\rvert \le 0.2\pi$ **Stopband:** $\lvert H(e^{j\omega})\rvert \le 0.2 \quad 0.6\pi \le \lvert\omega\rvert \le \pi$	10	2

5. Attempt any *one* part of the following:

a.	Design a low pass digital filter using Kaiser window satisfying the specifications given below: 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
b.	Explain the following terms with respect of finite word length effect in digital filters: (i) Coefficient quantization error, (ii)Quantization noise – truncation and rounding	10	3

6. Attempt any *one* part of the following:

a.	Given two sequences $x_1(n) = \{1, 2, 2\}$ and $x_2(n) = \{1, 2, 3, 4\}$. Determine the circular convolution of x1(n) and x2(n) using: i. Graphical Method ii. Stockholm's Method	10	4
b.	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.	10	4

7. Attempt any *one* part of the following:

a.	Briefly explain the applications of MDSP: Sub band Coding of Speech signals and Quadrature mirror filters with suitable diagram.	10	5
b.	Write the short note on: (i) Recursive Least Square Algorithm (ii) Window LMS Algorithm	10	5