Course Code: EEE3223

O P JINDAL UNIVERSITY

B. Tech. VI Semester Backlog Examination

DIGITAL SIGNAL PROCESSING

(Offered to EEE Program Code: 01UG030)

Time: 3 Hrs. **Max. Marks: 100**

Note: i) Part A is compulsory.

ii)Answer any one question from each unit of Section B

iii) All questions carry equal marks

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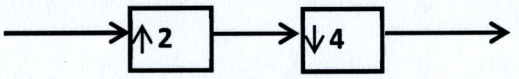
Section-A					
1	a.	Define IIR and FIR system.	2	3	1
	b.	What are causal systems?	2	2	2
	c.	What are the advantages of digital signal processing over analog signal processing?	2	5	2
	d.	Difference between discrete Fourier transform (DFT) and discrete-time Fourier transform (DTFT).	2	1	4
	e.	Define interpolation technique.	2	2	1
	f.	What are the properties that should have maintained for mapping from s-plane to z-plane?	2	1	2
	g.	Describe the canonic and non-canonic form of filter realization.	2	3	2
	h.	What is anti-aliasing filter?	2	4	1
	i.	What is a digital signal processor?	2	6	1
	j.	What are the necessary conditions for linear phase FIR filters?	2	3	2
Section-B: Unit-I					
2	a.	Let $x_1(n)$ and $x_2(n)$ be the flowing two 4-points sequences: $x_1(n)=\{1,2,2,1\}$ $x_2(n)=\{1,-1,-1,1\}$ Determine their linear convolution and linear convolution output.	8	2	3
	b.	Discuss the classification of systems.	8	1	2
OR					
3	a.	Compute the convolution $y(n)=x(n)\otimes h(n)$, where $x(n)=\{0,1,2,3,0\}$ and $h(n)=\{0,1,2,-2,-1,0\}$	8	1	3
	b.	Discuss the linear and circular convolution in detail.	8	2	2

Unit-II					
4	a.	Explain the fast Fourier transform in detail.	8	3	2
	b.	Obtain DFT of the following sequences: $x(n)=\{1,2,3,4,4,3,2,1\}$ using DIT FFT algorithm.	8	3	3
OR					
5	a.	Find the inverse DFT of $X(k)=\{3,2+j,1,2-j\}$.	8	3	3
	b.	Let x(n) be a finite duration sequence of length 8 such that $x(n)=\{-1,0,2,0,-4,0,2,0\}$. Find X(k) using DIT FFT flow graph.	8	2	2
Unit-III					
6	a.	Given the third order IIR system function $H(z)=\frac{0.2z^{2}+0.3z+0.04}{0.75z^{3}+0.6z^{2}+0.1z^{1}-0.4}$ Realize using direct form I and direct form II.	8	3	3
	b.	Realize the given FIR system function using i) direct form and ii) linear phase form $H(z)=\frac{1}{2}+\frac{1}{3}z^{-1}+z^{-2}+\frac{1}{4}z^{-3}+z^{-4}+\frac{1}{3}z^{-5}+\frac{1}{2}z^{-6}$	8	3	3
OR					
7	a.	Convert the given analog filter with a transfer function $H(s)=\frac{2}{(s+1)(s+3)}$ into a digital IIR filter using bilinear transformation. Assume T =0.1 sec.	8	3	2
	b.	A low-pass filter is to be designed with the following desired frequency response $H(\omega)=\begin{cases}e^{-j2\omega}, & \lvert\omega\rvert\le\frac{\pi}{4}\\ 0, & \frac{\pi}{4}<\lvert\omega\rvert\le\pi\end{cases}$ Determine the filter coefficient $h_d(n)$ if the window function is defined as $\omega(n)=\begin{cases}1, & 0\le n\le 4\\ 0, & otherwise\end{cases}$ Also, determine the frequency response $H(e^{j\omega})$ of the designed filter.	8	3	3
Unit-IV					
8	a.	Explain the decimation and interpolation process with suitable example. Also discuss why and where low-pass filter is used in both the processes.	8	4	4
	b.	What is meant by cascading of sampling rate converters? Discuss in detail with suitable examples.	8	4	2
OR					

9	a.	Explain the decimation in detail for the multirate digital signal processing system. Also discuss why and where low-pass filter is used.	8	4	4
	b.	For the given structure  The input is x(n) = {1, 3, 2, 5, -1, -2, 2, 3, 2, 1},the output y(n) will be?	8	4	3
UNIT-V					
10	a.	What are the factors the influences the selection of a DSP processor?	8	5	4
	b.	Discuss the different applications of programmable digital signal processors (PDSPs).	8	5	4
OR					
11	a.	Differentiate between Von Neumann architecture and Harvard architecture of processor.	8	6	2
	b.	Discuss the pipelining in DSP processor	8	5	2