

UNIVERSITY OF PETROLEUM AND ENERGY STUDIES
End Semester Examination, December 2021

Course: Digital Signal Processing
Program: B Tech ECE
Course Code: ECEG2013

Semester: V
Duration: 03 hrs.
Max. Marks: 100

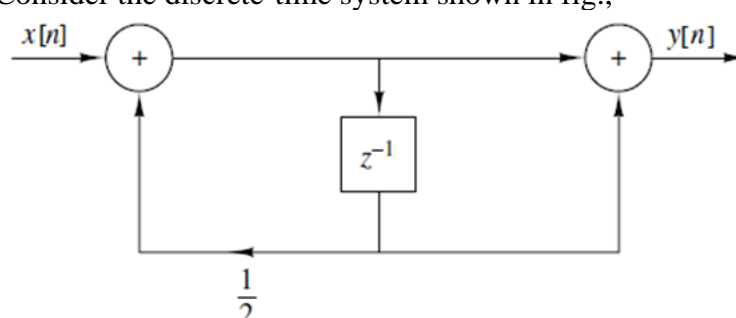
Instructions:

- Attempt all questions as per the instruction.
- Assume any data if required and indicate the same clearly. Use Table 1 data, in case if it is required.
- Unless otherwise indicated symbols and notations have their usual meanings.
- Strike off all unused blank pages

SECTION A (5Qx4 = 20 Marks)

S. No.		Marks	CO
Q 1	Define causality and stability of LTI discrete-time system with mathematical relations.	4	CO1
Q 2	Given sequence $x[k] = \begin{cases} 2; & k = 0,1,2 \\ 1; & k = 3,4 \\ 0; & otherwise \end{cases}$ Sketch the sequence $x[k]$ and the reverse sequence $x[-k]$, the shifted sequences $x[-k + 2]$ and $x[-k - 3]$.	4	CO1
Q 3	Define the sampling theorem. Find the DFT of the signal $x[n] = 3\delta[n] + 2\delta[n-1] + 3\delta[n-2]$ and draw the DFT spectrum $X[k]$.	4	CO2
Q 4	(a) Draw the block diagram of the system that represented by the following difference equation: $y[n] = b_0x[n] + b_1x[n - 1] + a_1y[n - 1]$ (b) Draw the magnitude characteristics of Chebyshev low pass filter (Type-I and Type-II) and label the specifications on its magnitude response plot.	2+2	CO3
Q 5	Digitize the analog filter with transfer function $H(s) = \frac{s+1}{s(s+2)}$ using the impulse invariant method, Assume the sampling frequency of 10 Hz	4	CO4

SECTION B (4Qx10 = 40 Marks)

Q 6	<p>Consider the discrete-time system shown in fig.,</p>  <p>(a) Find the input-output relation</p>	10	CO1
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	<p>(b) Compute the first 8-samples of its impulse response.</p> <p>(c) Check whether the given system is causal and stable.</p>		
Q 7	<p>(a) If $X[k]$ is the 5-point DFT of the sequence $x[n] = 2\delta[n] + \delta[n - 1] + \delta[n - 3]$. What sequence $y[n]$ has a 5-point DFT</p> $Y[k] = 2X[k] \cos\left(\frac{6\pi k}{10}\right)$ <p>(b) Compute 4-point DFT of the signal $x[n] = \{1, 2, -2, 1\}$ using decimation in time FFT algorithm.</p>	5+5	CO2
Q 8	<p>(a) Discuss in detail about frequency transformations.</p> <p>(b) Given the second-order IIR filter,</p> $H(z) = \frac{1 - 0.9z^{-1} + 0.1z^{-2}}{1 + 0.3z^{-1} + 0.04z^{-2}}$ <p>Realize $H(z)$ and develop difference equations using the following forms:</p> <p>(i) Direct form I</p> <p>(ii) Direct form II</p> <p>(iii) Cascade (series) form via the first-order sections.</p> <p>(iv) Parallel form via the first-order sections</p>	5+5	CO3
Q 9	<p>Design a 5-tap FIR band pass filter with a lower cutoff frequency of 1,600 Hz, an upper cutoff frequency of 1,800 Hz, and a sampling rate of 8,000 Hz using</p> <p>(a) Rectangular window function</p> <p>(b) Hamming window function.</p>	5+5	CO4
SECTION-C (2Qx20 = 40 Marks)			
Q 10	<p>A band pass filter with Butterworth magnitude-frequency response satisfies the following specifications:</p> <p>Passband: 0.3 – 3.4 kHz Stopband: 0 – 0.2 kHz and 4 – 8 kHz</p> <p>Pass band attenuation= 3 dB Stop band attenuation = 25 dB</p> <p>Sampling frequency = 16 kHz</p> <p>Obtain a suitable transfer function for the filter using the bilinear transformation method and realize the filter in direct form-I and II.</p> <p style="text-align: center;">OR</p> <p>Two linear phase FIR bandpass filters are required to satisfy the following specifications:</p> <p>For filter 1: passband: 8 – 12 kHz</p> <p style="padding-left: 40px;">Stopband ripple: 0.001</p> <p style="padding-left: 40px;">Peak passband ripple: 0.001</p> <p style="padding-left: 40px;">Sampling frequency: 44 kHz</p> <p style="padding-left: 40px;">Transition width: 3 kHz</p> <p>For filter 2: passband: 8 – 12 kHz</p> <p style="padding-left: 40px;">Stopband ripple: 0.01</p> <p style="padding-left: 40px;">Peak passband ripple: 0.001</p> <p style="padding-left: 40px;">Sampling frequency: 44 kHz</p> <p style="padding-left: 40px;">Transition width: 3 kHz</p>	20	CO3, CO4

	Obtain and compare the frequency response for each filter using the window method		
Q 11	<p>In a speech recording system with a sampling rate of 10,000 Hz, the speech is corrupted by broadband random noise. To remove the random noise while preserving speech information, the following specifications are given: Speech frequency range: 0 – 3,000 kHz Stopband range: 4,000 –5,000 Hz Passband ripple = 0.1 dB; Stopband attenuation = 60 dB</p> <p>(a) Design the FIR filter to remove random noise with the above specifications using Blackmann’s Window method. (b) Determine the difference equation and realize the FIR filter with suitable structure.</p>	20	CO3, CO4

Table 1: Prototype Lowpass filter Functions

Filter order (N)	3dB Butterworth Prototype Functions $H_p(s)$	Chebyshev Prototype Functions with 1dB Ripple $H_p(s)$
1	$\frac{1}{s + 1}$	$\frac{1.9652}{s + 1.9652}$
2	$\frac{1}{s^2 + 1.4142s + 1}$	$\frac{0.9826}{s^2 + 1.0977s + 1.1025}$
3	$\frac{1}{s^3 + 2s^2 + 2s + 1}$	$\frac{0.4913}{s^3 + 0.9883s^2 + 1.2384s + 0.4913}$
4	$\frac{1}{s^4 + 2.6131s^3 + 3.4142s^2 + 2.6131s + 1}$	$\frac{0.2456}{s^4 + 0.9368s^3 + 1.4539s^2 + 0.7426s + 0.2756}$
5	$\frac{1}{s^5 + 3.2361s^4 + 5.2361s^3 + 5.2361s^2 + 3.2361s + 1}$	$\frac{0.1228}{s^5 + 0.9368s^4 + 1.6888s^3 + 0.9744s^2 + 0.5805s + 0.1228}$