Code No: 156AR **R18**

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD
B. Tech III Year II Semester Examinations, February - 2023
DIGITAL SIGNAL PROCESSING
(Common to ECE, EIE)

Time: 3 Hours **Max. Marks: 75**

Note: i) Question paper consists of Part A, Part B.
ii) Part A is compulsory, which carries 25 marks. In Part A, Answer all questions.
iii) In Part B, Answer any one question from each unit. Each question carries 10 marks and may have a, b as sub questions.

PART – A

(25 Marks)

1.a) Define linear and non-linear systems. [2]
b) Give the properties of causal and stable systems. [3]
c) Define DFT pair. [2]
d) State and prove property time shifting of DFT. [3]
e) What are the properties of bilinear transformation? [2]
f) Discuss the impulse invariant method of designing IIR filter. [3]
g) In the design of FIR digital filter, how is Kaiser Window different from other windows? [2]
h) List out the advantages and disadvantages of FIR filters. [3]
i) What are the various basic building blocks used in realization of digital systems? [2]
j) Discuss about the frequency response of the stable systems. [3]

PART – B

(50 Marks)

2.a) Check for the stability and Causality of the following systems.
i) $y(n) = x(-n-3)$ ii) $y(n) = nx(n)$
b) Determine the linear convolution for the two sequences $x(n)=\{3,2,1,2\}$, $h(n)=\{1,2,1,2\}$. [4+6]

OR

3.a) Find impulse response of the system described by the difference equation $y(n)+ y(n-1)-2y(n-2)= x(n-1)+2x(n-2)$.
b) Discuss about the power signal and Energy signal. [6+4]

4.a) Find the output sequence $y(n)$ of a filter whose impulse response is $h(n)=[1,1,1]$; and input signal $x[n]=[3,-1,0,1,3,2,0,1,3,1]$ using overlap save method.
b) Explain use of FFT in linear filtering and correlation. [6+4]

OR

5.a) Compute the DFT of the following sequence using Radix -2, DIT-FFT algorithm $x[n] = [1, 1, 1, 1, 0, 0, 0, 0]$.
b) Describe Quantization errors in the direct computation of DFT. [6+4]

6.a) Explain the steps in designing an analog low pass chebyshev filter.

b) Using the bilinear transform, design a high pass filter, monotonic in pass band with cut off frequency of 100 Hz and down 10 dB at 350 H. the sampling frequency is 5000 Hz. [4+6]

OR

7. a) Obtain an Chebyshev filter transfer function that satisfies the constraints

$0.707 \leq |H(w)| \leq 1 \quad 0 \leq w \leq 0.2\pi$

$|H(w)| \leq 0.1 \quad 0.5\pi \leq w \leq \pi$

b) Give the advantages and disadvantages of the digital filters. [6+4]

8. Design a band pass filter which approximates the ideal filter with cutoff frequencies at 0.2 rad/sec and 0.3 rad/sec. The filter order is M=7. Use the Hanning window function. [10]

OR

9. a) Design a FIR low pass filter satisfying the following specifications $\alpha_p \leq 0.1$ dB; $\alpha_s \geq 44.0$ dB; $\omega_p = 20$ rad/sec; $\omega_s = 600$ rad/sec and $\omega_{sf} = 100$ rad/sec.

b) Explain different windows techniques in FIR digital filters. [6+4]

10. a) Realize the Direct form-I structure of the system given by

$$y[n] + 2\,y[n-1] + y[n-2] = x[n] + 0.75\,x[n-1]$$

b) Find the impulse response of the system described by difference equation

$$y(n) - 3y(n-1) - 4y(n-2) = x(n) + 2x(n-1)$$ using z transform. [5+5]

OR

11. a) Realize following digital filter by using direct form – II realization.

$$y(n) = 0.5y(n-1) - 0.25y(n-2) + x(n) + x(n-1)$$

b) Determine the impulse response of LTI system described by [5+5]

$$y(n) = x(n) + 0.9\,y(n-1) - 0.81y(n-2)$$

Code No: 156AR **R18**

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD
B. Tech III Year II Semester Examinations, February/March - 2022
DIGITAL SIGNAL PROCESSING
(Common to ECE, EIE)

Time: 3 Hours **Max. Marks: 75**

Answer any five questions
All questions carry equal marks
- - -

1.a) What are the conditions for stability and causality of an LTI system? Explain.
b) Explain in detail the classification of discrete-time systems.
c) What is the need for multi-stage implementation of sampling rate converters? Explain with an example. [5+5+5]

2.a) Find 8-point DFT X(K) of the real sequence.
$x(n) = \{0.707, 1, 0.707, 0, -0.707, -1, -0.707, 0\}$ by using DIF radix-2 FFT
b) Find the N-point DFT of $x(n) = b^n \cos an$ using the linearity property. [8+7]

3.a) State and prove any two properties of Discrete Fourier series.
b) Given $x(n) = 2^n$ and N=8, find X(k) using DIT-FFT algorithm. [6+9]

4.a) Design a digital low pass filter using Chebyshev filter that meets the following specifications: Passband magnitude characteristics that is constant to within 1dB for recurrences below $\omega = 0.2\pi$ and stopband attenuation of atleast 15dB for frequencies between $\omega = 0.3\pi$ and π. Use bilinear transformation.
b) Derive the relation between digital and analog frequencies in bilinear transformation. [10+5]

5.a) Design a Butterworth analog high pass filter that will meet the following specifications
i) Maximum pass band attenuation = 2dB
ii) Passband edge frequency = 200rad/sec
iii) Minimum stopband attenuation=20dB
iv) Stop band edge frequency = 100 rad/sec.
b) Prove that for a linear phase FIR filter the impulse response is symmetric. [8+7]

6.a) Explain the type II frequency sampling method of designing an FIR digital filter.
b) Design a band pass filter which approximates the ideal filter with cutoff-frequencies at 0.2rad/sec and 0.3rad/sec. The filter order is M=7. Use the Hanning window function. [5+10]

7.a) Explain coefficient quantization of IIR filters.
b) What is Round-off Noise in IIR Digital Filters? Discuss its effects in IIR filters. [7+8]

8.a) Describe various Structures of IIR filters with suitable diagrams.
b) Explain the limit cycle oscillations due to product round-off and overflow errors.[10+5]

R18

Code No: 156AR

JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD
B. Tech III Year II Semester Examinations, August/September - 2021
DIGITAL SIGNAL PROCESSING
(Common to ECE, EIE)

Time: 3 Hours **Max. Marks: 75**

Answer any five questions
All questions carry equal marks

- - -

1.a) Calculate the total response of the system described by
$y(n)-4y(n-1)-12y(n-2)=x(n)$, $y(-1)=1$, $y(-2)=2$.
b) Calculate the transfer function of the system defined by $y(n)-2y(n-1)=x(n)$. [10+5]

2.a) Describe with mathematical equations, how sampling rate can be decreased by a factor of D.
b) Briefly introduce the concepts of Multirate Digital Signal Processing. [10+5]

3.a) Derive the following properties of DFS.
i) Time shifting
ii) Time reversal
iii) Convolution.
b) Draw the butterfly diagram for DITFFT algorithm. [10+5]

4. Calculate the 8 point DFT of the sequence $x(n)=\{1,-2,3,1,-1,2\}$ using DIF-FFT and DIT-FFT. [15]

5.a) Write the differences between bilinear transform and impulse invariant method.
b) Write the differences between analog and digital filters. [8+7]

6. Design butterworth high pass filter for the given specifications:
$\alpha_p=3dB$, $\alpha_s=15dB$, $\Omega_p=1000\,rad/\sec$, $\Omega_s=500\,rad/\sec$. [15]

7. Given the filter specifications as

$$H_d(e^{j\omega})=e^{-j2\omega} \quad for\ 0\le|\omega|\le\frac{\pi}{2}$$
$$=0 \quad \frac{\pi}{2}\le|\omega|\le\pi$$

using rectangular window, calculate causal impulse response coefficients. [15]

8.a) Realize the following system equation in direct form-I and direct -form II

$$y(n)+3/4y(n-1)=x(n)-2x(n-1)$$

b) Write the differences between direct form-I and canonical form. [10+5]

Code No: RT32042 **R13** SET - 1

III B. Tech II Semester Supplementary Examinations, April/May -2019

DIGITAL SIGNAL PROCESSING

(Electronics and Communication Engineering)

Time: 3 hours Max. Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**
2. Answering the question in **Part-A** is compulsory
3. Answer any **THREE** Questions from **Part-B**

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PART –A

1	a)	What are the conditions for stability and causality of an LSI system?	[4M]
	b)	Define DFT and IDFT	[3M]
	c)	Find the z transform of x[n] = sins [ω_o n] u[n].	[4M]
	d)	What is the necessary and sufficient condition for linear phase Characteristics of an FIR filter?	[4M]
	e)	What is meant by aliasing? How to avoid it?	[3M]
	f)	What are the advantages of VLIW architecture?	[4M]

PART -B

2	a)	Determine the frequency response, and time delay of the systems given by $y(n)-\frac{1}{2}y(n-1)=x(n)$	[8M]
	b)	What is the significance of convolution? Explain	[8M]
3	a)	Compute the DFTs of the sequence $x(n)=2^{-n}$, where N = 8 using DIT algorithm.	[8M]
	b)	State any four properties of DFS and prove them	[8M]
4		Realize the following IIR system in the direct form I, direct from II and parallel forms. $H(z)=1/(1+az^{-1})(1-bz^{-1})$	[16M]
5	a)	The desired frequency response of a low pass filter is $H_d(e^{jw}) = \begin{cases} 1; \frac{-\pi}{2} \le w \le \frac{\pi}{2} \\ 0; \frac{\pi}{2} \le w \le \pi \end{cases}$ Determine h_d *(n) for M = 7* using a rectangular window	[8M]
	b)	Explain FIR filter design using windowing method.	[8M]
6	a)	Explain the following terms: i) Decimation ii) interpolation.	[8M]
	b)	What are the applications of Multi rate system? Explain.	[8M]
7	a)	What are the various addressing modes used in the TMS320C5X processor?	[8M]
	b)	What are the limitations of pipelining in Digital Signal Processor?	[8M]

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Code No: RT32042 **R13** SET - 1

III B. Tech II Semester Supplementary Examinations, November -2018

DIGITAL SIGNAL PROCESSING

(Electronics and Communication Engineering)

Time: 3 hours Max. Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**
2. Answering the question in **Part-A** is compulsory
3. Answer any **THREE** Questions from **Part-B**

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PART –A

1 a) Test the given system for time invariance : $y(n) = n\,x(n)$. [3M]

b) State any four properties of DFT [4M]

c) Find the Z-transform of $x(n) = (1/8)^n\ u(n)$ and its ROC. [4M]

d) Draw the direct form structure of $y(n) = \sum_{k=0}^{N-1} h[k]x[n-k]$ [4M]

e) What is the significance of Multirate Signal processing? What are the applications [3M]

f) What are the differences between fixed point processors and floating point Processors? [4M]

PART -B

2 a) Find the solution to the following linear constant coefficient difference equation with initial conditions $y(-1) = 4$ *and* $y(-2) = 10$ [8M]

$$y(n) - \frac{3}{2}y(n-1) + \frac{1}{2}y(n-2) = \left(\frac{1}{2}\right)^n \quad for\ n \geq 0$$

b) Explain the frequency domain representation of Discrete time signals [8M]

3 a) Given $x(n)=\{1,2,3,4,4,3,2,1\}$, find $X(k)$ using DIF FFT algorithm. [8M]

b) State and prove time – shifting and time scaling property of DFT. [8M]

4 a) Determine the ZT of x[n] = -na^n u [-n-1]. [8M]

b) What are the basic structures of FIR systems? Explain [8M]

5 a) What are the effects of windowing? Comparing various windowing techniques. [8M]

b) Design a High Pass FIR filter whose cut-off frequency is 1.2 radians/sec and N = 9 using Hamming Window. [8M]

6 a) Derive the frequency domain representation of decimator. [8M]

b) Explain the following terms: i) Up – sampling ii) Down- sampling [8M]

7 a) What is MAC? Explain its operation in detail. [8M]

b) Explain about Special addressing modes [8M]

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Code No: RT32042 **R13** SET - 1

III B. Tech II Semester Supplementary Examinations, April/May -2019

DIGITAL SIGNAL PROCESSING

(Electronics and Communication Engineering)

Time: 3 hours Max. Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**
2. Answering the question in **Part-A** is compulsory
3. Answer any **THREE** Questions from **Part-B**

*****

PART –A

1	a)	What are the conditions for stability and causality of an LSI system?	[4M]
	b)	Define DFT and IDFT	[3M]
	c)	Find the z transform of $x[n]$ = sins $[\omega_0 n]$ $u[n]$.	[4M]
	d)	What is the necessary and sufficient condition for linear phase Characteristics of an FIR filter?	[4M]
	e)	What is meant by aliasing? How to avoid it?	[3M]
	f)	What are the advantages of VLIW architecture?	[4M]

PART -B

2	a)	Determine the frequency response, and time delay of the systems given by $y(n)-\frac{1}{2}y(n-1)=x(n)$	[8M]
	b)	What is the significance of convolution? Explain	[8M]
3	a)	Compute the DFTs of the sequence $x(n)=2^{-n}$, where N = 8 using DIT algorithm.	[8M]
	b)	State any four properties of DFS and prove them	[8M]
4		Realize the following IIR system in the direct form I, direct from II and parallel forms. $H(z)=1/(1+az^{-1})(1-bz^{-1})$	[16M]
5	a)	The desired frequency response of a low pass filter is $H_d(e^{jw}) = \begin{cases} 1; \frac{-\pi}{2} \le w \le \frac{\pi}{2} \\ 0; \frac{\pi}{2} \le w \le \pi \end{cases}$ Determine $h_d(n)$ *for* $M = 7$ using a rectangular window	[8M]
	b)	Explain FIR filter design using windowing method.	[8M]
6	a)	Explain the following terms: i) Decimation ii) interpolation.	[8M]
	b)	What are the applications of Multi rate system? Explain.	[8M]
7	a)	What are the various addressing modes used in the TMS320C5X processor?	[8M]
	b)	What are the limitations of pipelining in Digital Signal Processor?	[8M]

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Code No: RT32042 **R13** SET - 1

III B. Tech II Semester Regular/Supplementary Examinations, April - 2017

DIGITAL SIGNAL PROCESSING

(Electronics and Communication Engineering)

Time: 3 hours Maximum Marks: 70

Note: 1. Question Paper consists of two parts (**Part-A** and **Part-B)**
2. Answering the question in **Part-A** is compulsory
3. Answer any **THREE** Questions from **Part-B**

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PART –A

1	a)	Test whether the following signal is periodic or not ,if periodic find the fundamental period $\sin\sqrt{2}\,\pi\, t$	[4M]
	b)	Find the DFT of a sequence x(n) ={1 ,1, 2, 2}	[4M]
	c)	Give block diagram representation of linear constant-coefficient difference equations.	[4M]
	d)	By impulse invariant method obtain the digital filter transfer function and the differential equation of the analog filter h(s) =1/s+1	[4M]
	e)	What are the applications of multi rate DSP?	[3M]
	f)	List special feature of DSP architecture.	[3M]

PART -B

2	a)	Determine whether each of the following systems defined below is (i) casual (ii) linear (iii) dynamic (iv) time invariant (i) $y(n) = \log_{10}[\{x(n)\}]$ (ii) y(n) = x(-n-2) (iii) y(n) = cosh[nx(n) + x(n-1)]	[12M]
	b)	Give the frequency domain representation of discrete time signals.	[4M]
3	a)	Compute the DFT for the sequence {1, 2, 0, 0, 0, 2, 1, 1}. Using radix -2 DIF FFT and radix -2 DIT- FFT algorithm.	[8M]
	b)	Derive the equation to implement a butterfly structure In DITFFT algorithm.	[8M]
4	a)	Realize the filter $H(z)=(z^{-1}-a)(z^{-1}-b)\ /\ (1-az^{-1})(1-bz^{-1})$ in cascade and parallel forms.	[8M]
	b)	State and prove time convolution property of Z-Transforms.	[8M]
5	a)	Obtain the impulse response of digital filter to correspond to an analog filter with impulse response $h_a(t) = 0.5\, e^{-2t}$ and with a sampling rate of 1.0kHz using impulse invariant method.	[8M]
	b)	Compare bilinear transformation and impulse invariant mapping.	[8M]
6	a)	Explain the decimation and interpolation process with an example. Also find the spectrum.	[8M]
	b)	The sequence x(n)=[0,2,4,6,8] is interpolated using interpolation sequence b_k =[1/2,1,1/2] and the interpolation factor is 2.find the interpolated sequence y(m).	[8M]
7	a)	Describe the multiplier/adder unit of TMS320c54xx processor with a neat block diagram.	[8M]
	b)	What are interrupts? What are the classes of interrupts available in the TMS320C5xx processor?	[8M]