Examination Control Division 2076 Chaitra

	Exam.		Regular	
,	Level	BE	Full Marks	80
-	Programme	BCT	Pass Marks	32
	Year/Part	IV / I	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing (CT 704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt <u>All</u> questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Define even and odd type discrete time signals with suitable example. Plot the signal x[-2n+3] where $x[n] = \{1, 2, 0, -1, -3, -4\}$. [2+3]
- 2. Determine whether the following system are: [5]
- a) y[n] = x[-n] is time-invariant or not.
 - b) $y[n] = x[n^2]$ is linear or not.
- 3. Find the output of LTI system having input signal x[n] = u[n+1]-u[n-4] and impulse response $h[n] = (1/2)^n u[n-1]$. [6]
- 4. Define ROC of z-transform. Find inverse z-transform using partial fraction expansion of $X(z) = (z^4 + 5z^3 3z + 4)/(z^2 1.5z 1)$, ROC: |z| < 0.5. [2+6]
- 5. Draw the pole-zero in the z-plane for a system with poles at 0.45 ± j1.06 and zeroes at 0.58±j2.06. Also plot the magnitude response (not to the scale) of the system. [2+6]
- 6. Compute Lattice and Ladder coefficients and Draw lattice-ladder structure for given IIR system $H(z) = (0.5 2z^{-1} + 3z^{-2})/(1 0.5z^{-1} 0.7z^{-2} + 0.3z^{-3})$. [6+4]
- 7. Realize the given system in Cascade form of 2nd order section in signal flow graph representation. [4]
 - $H(z) = \{ (1 0.5z^{-1})(1 + 0.35z^{-1})(1 0.3e^{j2n\pi/5}z^{-1})(1 0.3e^{-j2n\pi/5}z^{-1}) \} / \{ (1 0.6e^{jn\pi/3}z^{-1})(1 + 0.5e^{j2n\pi/7}z^{-1})(1 + 0.5e^{-j2n\pi/7}z^{-1}) \}$
- 8. Design the FIR filter using suitable window for the specifications: [6] $0.899 \le |H(e^{j\omega})| \le 1$, for $|\omega| \le 0.2\pi$ $|H(e^{j\omega})| \le 0.01$, for $0.4\pi \le \omega \le \pi$
- 9. What is optimum filter? Show mathematical expression of Remez exchange algorithm for FIR filter design. [1+5]
- 10. Design a digital low pass Butterworth filter by applying bilinear transformation techniques for the given specifications:

 [10]

Passband peak to peak ripple ≤ 1dB

Passband edge frequency = 1.2KHz

Stopband Attenuation ≥ 40dB

Stopband edge frequency = 2.5 KHz

Sample rate = 8KHz

- 11. Find 8-point DFT of sequence x[n] = {1, 2, 3, 3, 5, 0, 4, 6} using Decimation in frequency Fast Fourier Transform (DIFFFT) algorithm. [7]
- 12. Find $x_3[n]$ if DFT of $x_3[n]$ is given by $X_3(k) = X_1(k) * X_2(k)$ where $X_1(k)$ and $X_2(k)$ are 4-point DFT of $x_1[n] = \{1, 2, -2\}$ and $x_2[n] = \{1, 2, 3, -1\}$ respectively. [5]

Examination Control Division 2075 Chaitra

Exam.	Regular / Back				
Level	BE	Full Marks	80		
Programme	BCT	Pass Marks	32		
Year / Part	IV / I	Time	3 hrs.		

[3+4]

Subject: - Digital Signal Analysis and Processing (CT 704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- ✓ The figures in the margin indicate *Full Marks*.
- ✓ Assume suitable data if necessary.
- 1. Define Power and Energy type discrete time signal with suitable example. Differentiate between Fourier Series and Fourier Transform.
- 2. Find the output of LTI system having impulse response h[n] with h[-2] = 3, h[0] = 2, h[1] = 1 and input signal $x[n] = (2)^n$, for $-1 \le n \le 3$. Also check the answer. [5+2]
- 3. Plot the pole-zero in z-plane and draw magnitude response (not to scale) of the system described by differential equation

$$y(n) - 0.3y(n-1) = 2x(n-2) + 0.7x(n-1) + 4x(n)$$
 [2+7]

4. Draw the lattice structure from the following system function

$$H(t) = \frac{1}{1 + \frac{2}{3}z^{-1} + \frac{5}{8}z^{-2} + \frac{2}{3}z^{-3} + z^{-4}}$$
 [9]

- 5. What is optimum filter? Show mathematical expression of Remez exchange algorithm for FIR filter design. [2+6]
- 6. List out the properties of Region of convergence and locate the ROC of the following signal

$$x[n] = (0.1)^n u[n] + (0.3)^n u[-n-1]$$
 [4+6]

7. Using bilinear transformation, design a digital filter using Butterworth approximation which satisfies the following conditions

$$0.8 \le |\text{He}^{JW}| \le 1 \text{ for } 0 \le W \le 0.2\Pi$$

 $|\text{He}^{JW}| \le 0.2 \text{ for } 0.6\Pi \le W \le \Pi$ [10]

- 8. How fast is FFT? Find X(3) and X(5) for given sequence $x[n] = \{1, -2, 3, 2\}$ using DITFFT algorithm. [2+8]
- 9. Differentiate between linear convolution and circular convolution compute circular convolution of signals

$$X_1[n] = \{0, 0, 1, 1\} \text{ and } X_2[n] = \{1, 1, 1, 1\}$$
 [3+7]

TRIBHUVAN UNIVERSITY

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Examination Control Division 2076 Ashwin

Exam.		Back	
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV/I	Time	3 hrs.

[4+3]

Subject: - Digital Signal Analysis and Processing (CT 704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Explain Fourier transform multiplication property for two sequences. Write Drichlet's conditions for Fourier series.
- 2. Find convolution between two signals $x[n] = 2^n 4[-n]$, 0 < a < 1 and h[n] = 4[n] [6]
- 3. State Convolution property of Z-transform. Find inverse Z-transform of

$$X(z) = z / \{(z - 0.6)(z + 0.5)^2\}, ROC: |z| > 0.6$$
 [3+6]

- 4. Describe stability and causality characteristics of LTI system in terms of Impulse Response and ROC of its transfer function with suitable examples. [4+3]
- 5. Compute Lattice and Ladder coefficients and Draw lattice-ladder structure for given IIR system $H(z) = (0.7 1.5z^{-1} + 0.5z^{-2}) / (1 0.5z^{-1} 0.7z^{-2} + 0.3z^{-3})$ [6+3]
- 6. For the system described by the following difference equation: [2+8]

$$y[n] = 0.67x[n] - 0.3x[n-1] + 2.75y[n-1]$$

Map the poles and zero in the z-plane and plot the phase response of the system.

7. Design a low pass discrete IIR filter by Bilinear Transformation method to an approximate Butterworth filter having specifications as below: [12]

Pass bandedge frequency (ω_p) = 0.22 π radians

Stop bandedge frequency (ω_s) = 0.54 π radians

Passband ripple $(\delta_p) = 0.11$

Stopband ripple $(\delta_s) = 0.22$, Consider sampling frequency 0.5 Hz.

- 8. Why we need DFT? Find 8-point DFT of sequence x[n] = {1, 2, 3, 3, 5, 1, 4, 2} using Decimation in frequency Fast Fourier Transform (DIFFFT) algorithm. [2+8]
- 9. In which case do we choose FIR filter and IIR filter? Design a Kaiser Window to meet the following specifications. [2+4+4]

$$0.99 \le \left| H(e^{jw}) \right| \le 1.01,$$
 for $0 \le w \le 0.16\pi$
 $\left| H(e^{jw}) \right| \le 0.01,$ for $0.18\pi \le w \le 2\pi$

Draw the flow chart for Remez- Exchange algorithm

Examination Control Division 2074 Chaitra

Exam.		Regular	
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV / I	Time	3 hrs.

[3+2]

[3+3]

[5]

[12]

[3+7]

Subject: - Digital Signal Analysis and Processing (CT704)

- Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- \checkmark The figures in the margin indicate <u>Full Marks</u>.
- ✓ Assume suitable data if necessary.
- 1. Plot the sequence $x[n] = u[n] u[n-3] + 5\delta[n-4] = nu[n-6]$. List out the properties of LTI system.
- 2. Determine whether the following system are:
 - a) y[n] = y[n-4] + x[n-4] is Time-invariant or not
 - b) $y[n] = x^2[n]$ is Linear or Nor-linear
- 3. Define a ROC. What are the properties of ROC of z-transform? Find the inverse Ztransform of $X(z) = (2z^2 + 2z^2 + 3z + 5)/(z^2 - 0.1z - 0.2)$, ROC: |z| < 0.4. [1+3+5]
- 4. The poles of a system are located at: 0.45-0.77i and -2±0.3i. Map the poles and zero in [2+8]the z-plane and plot the magnitude response of the system.
- 5. Obtain the Direct Form I and Direct Form II realization of the following system. [5]
 - 3y[n] + y[n-1] + 2y[n-4] = 2x[n] + x[n-3]
- 6. Determine the lattice coefficients coefficients corresponding to the FIR filter with the system function:

$$H(z) = A_3(z) = 1 + \frac{52}{96}z^{-1} + \frac{25}{40}z^{-2} + \frac{1}{3}z^{-3}$$

7. Design a digital low-pass filter with the following specification:

i) Pass-band magnitude constant to 0.7 dB below the frequency of 0.15π ii) Stop-band attenuation at least 14 dB for the frequencies between 0.6π to π Use Butter worth approximation as a prototype and use impulse invariance method to obtain the digital filter.

8. Design a FIR linear phase filter using Kaiser window that meets the following [9+3]specifications:

 $|H(e^{iw})| \le 0.01, 0 \le |w| \le 0.25\pi$ $0.95 \le |H(ejw)| \le 1.05, 0.35\pi \le |w| \le 0.6\pi$

 $|H(e^{jw})| \le 0.01, 0.65\pi \le |w| \le \pi$

Also determine the minimum length (M+1) of the impulse response and Kaiser window parameter β.

- 9. Why do we need DFT? Draw the butterfly structure to compute the DFT of the following signal using Radix-2 DIFFFT algorithm, and compute X(2) and X(1) only $x[n] = \{1.5, -1, 1.8, 0.6, 3, 1.7\}$
- 10. Define zero padding. Find the linear convolution through circular convolution with padding of zeros for the following sequences: $x[n] = \{1,1,1,1\}$ and $h[n]\{2,3\}$. [1+5]

Examination Control Division 2075 Ashwin

Exam.		Back		
Level	BE	Full Marks	80	
Programme	BCT	Pass Marks	32	
Year / Part	IV / I	Time	3 hrs.	

Subject: - Digital Signal Analysis and Processing (CT704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Determine whether the following sequences are linear or not:

[3+3]

[5]

- a) $y[n] = x^2[n]$
- b) $y[n] = \cos\left(\frac{5\pi}{8}n + \frac{\pi}{4}\right)$
- 2. Find the output of LTI system having impulse response $h[n] = 2^n * \{u[n] u[n-3]\}$ and input signal $x[n] = \delta[n] + \delta[n-1] + \delta[n-2]$.
- 3. List out the properties of Region of convergence and locate the ROC of the following signal. [3+6]

$$x[n] = (0.6)^n u[n] + (0.25)^n u[n]$$

- 4. Draw the poles and zeros in the z-plane for a system with poles at 0.45±j1.06 and zeros at 0.58±j2.06. Also plot the magnitude response of the system. [2+8]
- 5. Draw the Lattice structure from the following system function: [7+3]

$$\frac{1}{3 + \frac{39}{24}Z^{-1} + \frac{15}{8}Z^{-2} + \frac{3}{9}Z^{-3}}$$

And represent $\frac{5}{8}$ and $-\frac{5}{8}$ in sign magnitude, 1's complement and 2's complement format.

6. Design a digital low-pass filter with the following specification:

[12]

- i) Pass-band magnitude constant to 0.7 dB below the frequency of 0.15 π
- ii) Stop-band attenuation at least 14 dB for the frequencies between 0.6π to π

Use Butterworth approximation as a prototype and use bilinear transformation method to obtain the digital filter.

7. Design a linear phase FIR filter using Kaiser Window to meet the following specifications: [8+4]

$$0.99 \le |H(e^{jw})| \le 1.01,$$
 for $0 \le w \le 0.19\pi$
 $|H(e^{jw})| \le 0.01,$ for $0.21\pi \le w \le \pi$

Draw the flow chart for Optimum filter design.

- 8. How fast is FFT compare to DFT? Draw the butterfly diagram of 8-point DFT of a sequence as x[n]=n+1 using Decimation in Time FFT algorithm. [3+7]
- 9. State the circular convolution property of DFT. Find the circular convolution of: [1+5]

$$x_1(n) = \{1, 2-1, 1\}$$
 and $x_2(n) = \{1, 3, 5, 7\}$

Examination Control Division 2073 Chaitra

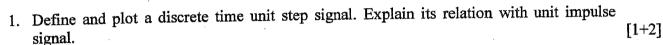
Exam.		Regular	
Level	BE	Full Marks	80
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[10]

[7]

Subject: - Digital Signal Analysis and Processing (CT704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.



- 2. Calculate y[n], if x[n] is x[-2] = 0.5, x[0] = 1, x[1] = 0.75, x[3] = 0.5 and x[0] = 1, x[1] = 0.75 and x[2] = 0.5 and verify your result. [6]
- 3. Define a ROC. Find inverse Z-transform of $X(z) = (2z^3+2z^2+3z+5) / (z^2-0.1z-0.2)$, ROC: |z| < 0.4
- 4. Define the difference equation with example. The Poles of a system are located at: 0.45 + 0.77i and $2 \pm 0.7i$ and zeros at: $1.2 \pm 0.43i$. Plot the magnitude response of this system. [2+8]
- 5. Draw the Lattice Structure from the following system function:

$$\frac{1 + \frac{1}{3}z^{-1} + \frac{9}{8}z^{-2} + \frac{4}{3}z^{-3} + z^{-4}}{1 + \frac{2}{3}z^{-1} + \frac{5}{8}z^{-2} + \frac{2}{3}z^{-3} + z^{-4}}$$

6. Design a digital Butterworth low pass filter satisfying the constraints

$$\begin{cases} 0.707 \le \left| H(e^{jw}) \right| \le 1 & 0 \le w \le \frac{\pi}{2} \\ \left| H(e^{jw}) \right| \le 0.2 & \frac{3\pi}{4} \le w \le \pi \end{cases}$$

With T = 1sec using bilinear transformation method. Realize the filter using the most convenient realization form. [11+4]

7. Design an FIR linear phase filter using Kaiser window to meet the following specifications: [8]

$$0.98 \leq \left| H(e^{jw}) \right| \leq 1.02, for \ 0 \geq w \geq 0.9\pi$$

$$|H(e^{jw})| \le 0.01, for \ 0.14\pi \le w \le \pi$$

- 8. Draw the Howchart of Remez-Exchange theorem and explain it.
- 9. Why we need FFT? Find 8-point DFT of sequence x[n] = {1, -1, 3, 2, 1, 1, 3, -2} using Decimation in frequency Fast Fourier Transform (DIFFFT) algorithm. [2+6]
- 10. Find $x_3[n]$ if DFT of $x_3[n]$ is given by $X_3(k) = X_1(k)$ $X_2(k)$ where $X_1(k)$ and $X_2(k)$ are 5-point DFT of $x_1[n] = \{1, -2, 5, 1, 2\}$ and $x_2[n] = \{1, 2, -3, -2\}$ respectively. [7]

Examination Control Division 2074 Ashwin

Exam.		Back	
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV / I	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing (CT704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- The figures in the margin indicate Full Marks.
- Assume suitable data if necessary.
- 1. Define Energy and Power type discrete time signal. Check whether signal $x[n] = e^{j(\pi n/3 + \pi/4)}$ is periodic or not. If it is periodic, state its periodic time. [2+2]
- 2. Find the output of LTI system having impulse response $h[n] = (1/2)^n \{u[n+2] u[n-2]\}$ and input signal $x[n] = \{2, 1, 0.5, -1\}$. Also check the answer. [3+2]
- 3. State and explain the properties of a Region of Convergence (ROC). Find the inverse z-transform of $X(z) = z^2 \left[1 \frac{3}{2}z^{-1}\right] (1 + z^{-1})(1 z^{-1})$ [3+3]
- 4. Plot the pole-zero in z-plane and Draw Magnitude Response (not to the scale) of the system described by difference equation y[n]=0.4y[n-1]+0.2y[n-2]=x[n]+0.5x[n-1]+0.6x[n-2]+0.8x[n-3] [3+7]
- 5. Draw the direct form and Lattice structure of a filter with system function $H(z) = 1 + 0.7z^{-1} + 1.2z^{-2} z^{-3}$. [3+7]
- 6. Why Kaiser window is better than other fixed windows in FIR filter design? Find out first six coefficients of impulse response of a low pass FIR filter having Pass band edge frequency $\omega_p = 0.2\pi$, Stop band edge frequency $\omega_s = 0.5\pi$ and Stop band attenuation $\alpha_s = 41 \text{dB}$ using any appropriate window function. [2+6]
- 7. What is an optimum filter? Show mathematical expression of the Remez exchange algorithm for FIR filter design with flow chart. [1+6]
- 8. Design a low pass discrete IIR filter by Bilinear Transformation method to an approximate Butterworth filter having specifications as below: [15]

Pass bandedge frequency $(\omega_p) = 0.27 \pi$ radians Stop bandedge frequency $(\omega_s) = 0.58 \pi$ radians

Passband ripple $(\delta_p) = 0.11$

Stopband ripple $(\delta_s) = 0.21$, Consider sampling frequency 0.5 Hz.

- 9. Compute the 8-point DFT of the sequence $x[n] = \left\{ \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0 \right\}$ using Decimation in Frequency Fast Fourier Transform (DIF-FFT) algorithm. [7]
- 10. What is a zero padding? If $X_1(k)$ and $X_2(k)$ are DFT of sequence $x_1[n] = \{1, 2, 0, 1, -2\}$ and $x_2[n] = \{1, 0, 1, 1, 2\}$ respectively then find the sequence $x_3[n]$; If DFT of $x_3[n]$ is given by $X_3(k) = X_1(k)$, $X_2(k)$. [1+7]

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Examination Control Division 2072 Chaitra

		75	
Exam.		Regular	
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
	IV/I	Time	3 hrs.

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- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt <u>All</u> questions.
- ✓ The figures in the margin indicate <u>Full Marks</u>.
- ✓ Assume suitable data if necessary.

1.	How fourier series coefficients are calculated? Explain.	[4]
2.	Find the output of LTI system having impulse response $h[n]$ with $h[-2] = 1$, $h[0] = 2$, $h[1] = 3$ and input signal $x[n]$ with $x[0] = 1/2$, $x[2] = 2$, $x[3] = 3$. Also check the answer.	[3+2]
3.	Explain the properties of Region of Convergence with examples.	[6]
4.	Describe stability and causality characteristics of LTI system in terms of Impulse Response and ROC of its transfer function with suitable examples.	[4]
5.	Plot the pole-zero in z-plane and Draw Magnitude Response (not to the scale) of the system described by difference equation.	[2+4]
	y[n] - 0.4y[n-1] + 0.1y[n-2] = x[n] + 0.6x[n-1]	. 1
6.	Determine the Direct Form I and Direct Form II realization of the following system.	[5]
	y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-2) + 0.6x(n-2)	
7.	Compute the lattice coefficients and draw the lattice structure of following FIR system.	[5]
	$H(z) = 1 + 2z^{-1} + z^{-2}$	
8.	Describe how digital FIR filter can be design by window method. Why Kaiser window is better than other fixed windows in FIR filter design?	[5+3]
9	What is an optimum filter? Show mathematical expression of Remez exchange algorithm for FIR filter design.	[1+6]
- 1	0. Explain about the advantages of selecting bilinear transformation method over impulse invariance method (I I M). Design a digital low pass Butterworth filter using impluse invariant transformation with pass band and stop band frequencies 200Hz and 500Hz respectively. The pass band and stop band attenuation are -5dB and -12dB respectively. The sampling frequency is 5kHz. Use IIM method.	
1	1. Find the FFT of the signal $x[n]\{1,1,2,4,3,1,2,1\}$ using DIT-FFT algorithm.	[8]
	2 Compute Circular Convolution of $h(n) = \{1, 2, 1, -1, 1\}$ and $x[n] = \{1, 2, 3, 1\}$.	[7]

Examination Control Division 2073 Shrawan

Exam.	New Back (2066 & Later Batch)				
Level	BE	Full Marks			
Programme	BCT	Pass Marks	32		
Year / Part	IV / I	Time	3 hrs.		

[3]

Subject: - Digital Signal Analysis and Processing (CT704)

- Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Explain the process of calculating fourier series coefficients. [3] 2. Determine the system output y(n) of the following signals: $h(n) = \{1,1,1\}$ and $x(n) = \{1,1,1,1\}$ [6] 3. Define a ROC. Find inverse Z-transform of $X(z) = z/\{(z-0.4)(z+1.5)^2\}$, ROC: |z| < 0.4[1+5] 4. State linear constant coefficient difference equation and corresponding system function. Determine the output sequence of the system with impulse response $h[n] = (1/2)^n u[n]$ when the input signal is $x[n] = 10 - 5\sin(\pi n/2) + 20\cos(\pi n) - \infty < n < \infty$. [3+7] 5. The system function of a filter is $H(z) = 2 + 1.8z^{-1} - 1.6z^{-2} + z^{-3}$. Draw the Direct Form and Lattice Structure implementation of the above filter. [3+7]6. Explain in detail about how rectangular window is used in FIR filter design. How Gibb's oscillations arise in this process. [6] 7. Explain about Remaz exchange algorithm with suitable derivation and flow chart. [9] 8. Using bilinear transformation, design a butterworth low pass filter which satisfies the following Magnitude Response. [12] $0.89125 \le |H(e^{jw})| \le 1$ for $0 \le \omega \le 0.2\pi$ $|H(e^{iw})| \le 0.17783$ for $0.3\pi \le \omega \le \pi$
- 9. Explain briefly about bilinear transformation method of IIR filter design.
- 10. Why do we need DFT? Find 8-point DFT of sequence $x[n] = \{1,-1,2,2,1,1,2,2\}$ using Fast Fourier Transform algorithm. [2+6]
- 11. Find $x_3[n]$ if DFT of $x_3[n]$ is given by $X_3(k) = X_1(k) X_2(k)$ where $X_1(k)$ and $X_2(k)$ are 5-point DFT of $x_1[n] = \{1,-2,2,1,4\}$ and $x_2[n] = \{2,1,-3,-1\}$ respectively. [7]

Examination Control Division 2072 Kartik

Exam. New Back (2066 & Later Batch)				
Level	BE	Full Marks	80	
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Year / Part	IV / I	Time	3 hrs.	

[6]

[15]

[7]

Subject: - Digital Signal Analysis and Processing (CT704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Define energy and power signal. Check the signal x[n] = u[n] and $x[n] = \delta[n]$ is Energy or Power type. [2+3]
- 2. Find the output of LTI system having impulse response $h[n] = (1/3)^n \{u[n+1]-u[n-2]\}$ and input signal $x[n] = \{2,1,0.5,3\}$. [5]
- 3. State the properties of region of convergence (ROC). Drive the convolution property of Z-transform. [3+3]
- 4. Find the output of LTI System having impulse response $h[n] = (1/2)^n u[n]$ and input signal $x[n] = 5e^{j\pi n/3}$ for $-\infty < n < \infty$. [4]
- 5. Plot Magnitude Response (not to the scale) of the system described by difference equation.

y[n]-0.5y[n-1]+0.3y[n-2] = x[n]+0.7x[n-1]

- 6. Determine the Direct Form II realization of the following system y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) 0.252x(n-2)
- 7. Compute the lattice coefficients and draw the lattice structure of following FIR system [6] $H(z) = 1 + 2z^{-1} 3z^{-2} + 4z^{-3}$
- 8. Draw the flowchart of Remez-Exchange theorem and explain it. Design an FIR linear phase filter using Kaiser window to meet the following specifications: [6+8]

 $0.99 \le |H(e^{jw})| \le 1.01$, for $0 \ge w \ge 0.19\pi$

 $\left| H(e^{jw}) \right| \le 0.01$, for $0.21\pi \le w \le \pi$

9. Design a low pass digital filter by Bilinear Transformation method to an approximate Butterworth filter, if passband edge frequency is 0.25π radians and maximum deviation of 1 dB below 0 dB gain in the passband. The maximum gain of -15 dB and frequency is 0.45π radians in stopband, Consider sampling frequency 1Hz.

10. Find 8-point DFT of sequence $x[n] = \{1,1,0,1,0,1,2\}$ using Decimation in Time Fast Fourier Transform (DITFFT) algorithm.

11. Why we need DFT? If $X_1(k)$ and $X_2(k)$ are DFT of sequence $x_1[n] = \{1,2,4\}$ and $x_2[n] = \{-1,2,3,1\}$ respectively, then find the sequence $x_3[n]$, if DFT of $x_3[n]$ is given by $X_3(k) = X_1(k) X_2(k)$. [2+6]

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Examination Control Division 2071 Chaitra

Exam.	and the second s	Regular	
Level	BB	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	ĪV / I	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing (CT704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt <u>All</u> questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- Find the even and odd part of signal x[n],

 $x[n] = \begin{cases} 1 & \text{for } -4 \le n \le 0 \\ 2 & \text{for } 1 \le n \le 4 \end{cases}$

- 2. A discrete time LTI system has impulse response $h(n) = \{1,3,2,-1,1\}$ for $-1 \le n \le 3$.

 Determine the system output y(n) if the input x(n) is given by $x(n) = 2\delta(n) \delta(n-1)$. [6]
- 3. Define ROC. Find inverse Z-transform of

 $X(z)=1/\{(z-0.5)(z+2)\}$, if

- i) ||ROC: 0.5 < |z| < 2
- ii) ROC: |z| < 0.5
- iii) ROC: $|\mathbf{z}| > 2$
- 4. The poles of a system are located at: $0.45\pm0.77i$ and $-2\pm0.3i$ and zeroes at: $1.2\pm3i$. Map the poles and zero in the z-plane and plot the magnitude response of the system. [2+8]
- 5. Compute Lattice coefficients and draw lattice structure for given IIR system $H(z) = 1/(1-0.01z^{-1} 0.23z^{-2} + 0.5z^{-3})$. Also check the stability of given system. [4+2+1]
- 6. What is limit cycle effect in recursive system? Describe with one example showing how it occurs. [3]
- 7. Design a low pass FIR filter having Pass band edge frequency $\omega_p = 0.3\pi$, Stop band edge frequency $\omega_s = 0.5 \pi$ and Stop band attenuation $\alpha_s = 40$ dB using any appropriate window function. [8]
- 8. What is optimum filter? Show mathematical expression of Remez exchange algorithm for FIR filter design. [1+6]
- 9. What is the advantage of bilinear transformation? Design a low pass discrete time
 Butterworth filter applying bilinear transformation having specifications as follows: [2+9+4]

Pass band frequency $(w_p) = 0.25 \pi$ radians Stop band frequency $(w_s) = 0.55 \pi$ radians

Pass band ripple $(\delta_p) = 0.11$

and stop band ripple (δ_s) = 0.21

Consider sampling frequency 0.5 Hz.

Also, convert the obtained digital low-pass filter to high-pass filter with new pass band frequency $(w'_P) = 0.45 \pi$ using digital domain transformation.

- 10. Why do we need Discrete Fourier Transform (DFT) although we have Discrete-time Fourier Transform (DTFT)? Find circular convolution between $x[n] = \{1, 2\}$ and y[n] = u[n] u[n-4].
- 11. How fast is FFT? Draw the butterfly diagram and compute the value of X(7) using 8 pt DIT-FFT for the following sequences: $x(n) = \{1, 0, 0, 0, 0, 0, 0, 0\}$

[2+6]

[2+5]

[3]

[1+5]



Examination Control Division 2071 Shawan

Exam.			LATE DAY
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV/I	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing (CT704)

✓ Candidates are required to give their answers in their own words as far as practicable.

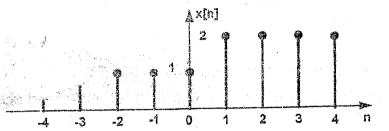
✓ Attempt <u>All</u> questions.

✓ The figures in the margin indicate <u>Full Marks</u>.

✓ Assume suitable data if necessary.

1. Find the odd and even part of the following signal:

[4+5]



A discrete time LTI system has input signal and impulse response as,

 $x[n] = \begin{cases} 1 & -1 \le n \le 1 \\ 0 & elsewhere \end{cases}$ and $h[n] = \begin{cases} 1 & -1 \le n \le 1 \\ 0 & elsewhere \end{cases}$ Find the output of the system using graphical method.

2. Find the inverse z transform of:

[6]

$$X(Z) = (1+2z^{-1}+z^{-2})/(1+1.5z^{-1}+0.5z^{-2}), |z| > 1$$

using partial fraction method.

3. Why do we need difference equation? State linear constant coefficient difference equation and corresponding system function.

[2+3+5]

Consider an LTI system with impulse response h[n]=(1/2)ⁿ u[n]. Determine y[n], if the input is x[n] = Ae^{ink}

4. If a 3 stage lattice filter for all pole polynomial has coefficients.

[5]

$$K_1 = \frac{1}{4}$$
, $K_2 = \frac{1}{2}$ and $K_3 = \frac{1}{3}$ Obtain the system function of this filter.

- 5. What is the importance of quantization in Digital Signal Processing? Which one is better rounding or truncation? Explain about limit cycles in recursive system? Define dead band. [1+1+2+1]
- 6. Explain in detail about how rectangular window is used in FIR filter design. How Gibb's oscillations arise in this process.

[6]

7. What is a Remez exchange algorithm? Derive its equation and draw its flow chart.

[9]

8. Design a low pass digital filter by Bilinear Transformation method to an approximate Butter worth filter it passband frequency is 0.2π radians and maximum deviation of 1 db below 0 dB gain in the pass band. The maximum gain of -15 db and frequency is 0.4π radians in stop band, consider sampling frequency 1 Hz.

[15]

9. A system has input signal $x[n] = \{1,2,3,4\}$ and impulse response $h[n] = \{1,3,5,7\}$ and the DFT of x[n] is X[k] and the DFT of h[n] is H[k]. Find the output of the system y[n] if G[k] = X[k].H[k]

[7]

10. Find DFT for {1, 1, 2, 0, 1, 2, 0, 1} using FFT DIT butterfly algorithm and plot the spectrum.

[6+2]

Examination Control Division 2070 Chaitra

Exam.		Regular	
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV/I	Time	3 hrs.

[3]

[2+4]

Subject: - Digital Signal Analysis and Processing (CT704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt All questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Determine which of the following signals are periodic and compute their fundamental period:
 - i) $Cos(\pi tn^2/8)$
 - ii) $Cos(n/2) cos(\pi n/4)$
- 2. Find output, y(n) when: $h(n) = \{5,4,3,2\}$ and $x(n) = \{1,0,3,2\}$ [6]
- 3. List out the properties of Region of Convergence. Find the Z-transform and locate the ROC of the signal.

$$x[n] = \left(-\frac{1}{3}\right)^n u[n] - \left(\frac{1}{3}\right)^n u[-n-1]$$

- 4. Find the output of LTI System having impulse response [4] $h[n] = (1/3)^n u[n]$ and input signal $x[n] = 5e^{j\pi n/2}$ for $-\infty < n < \infty$.
- 5. Plot Magnitude Response (not to the scale) of the system described by difference equation, y[n] 0.3 y[n-1] + 0.225y[n-2] = x[n] + 0.5x[n-1] [6]
 - 6. Determine the Cascade Form realization of the following system. [4] $y[n] \frac{3}{4}y[n-1] + \frac{1}{8}y[n-2] x[n] 2x[n-1] = 0$
 - 7. Compute the lattice coefficients and draw the lattice structure of following FIR system $H(z) = 1 + 3.1z^{-1} + 5.5z^{-2} + 4.2z^{-3} + 2.3z^{-4}$ [6]
 - 8. Describe how FIR filter can be designed by window method. Discuss the characteristics of different type of window function. [4+4]
 - 9. What is an optimum filter? Show mathematical expression of Remez exchange algorithm for FIR filter design. [1+6]
 - 10. Using bilinear transformation method, design a digital filter using Butterworth approximation which satisfiers the following conditions: [10]

$$0.8 \le \left| \text{He}^{\text{jw}} \right| \le 1$$
 for $0 \le \text{w} \le 0.2\pi$
 $\left| \text{He}^{\text{jw}} \right| \le 0.2$ for $0.6\pi \le \text{w} \le \pi$

- 11. A digital LPF with cut off frequency $w_c = 0.2575 \pi$ is given as $H(Z) = \frac{0.1 + 0.4z^{-1}}{1 0.6z^{-1} + 0.1z^{-2}}$ Design a digital high pass filter with $w'_c = 0.3567\pi$. [5]
- 12. Define Padding zones. Find 8-point DFT of sequence.

 x[n] = {1,1,0,0,1,1,2} using Decimation in Time Fast Fourier Transform (DITFFT) algorithm.

 [1+6]
- 13. Why we need DFT? State and prove Circular Convolution property of DFT. [2+2+4]

Examination Control Division 2070 Ashad

Exam.	New Back (2	066 & Later	Batch)
Level	BE	Full Marks	80
Programme		Pass Marks	32
Year / Part	IV / I	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing (CT704)

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt <u>All</u> questions.
- ✓ The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Find the even and odd part of signal x[n],

 $x[n] = \begin{cases} 1 & \text{for } -4 \le n \le 0 \\ 2 & \text{for } 1 \le 4 \end{cases}$ [3]

- 2. Illustrate the significance of convolution summation in digital signal analysis. Compute the convolution of the following signals: $h(n) = \{1,0,1\}$ and $x(n) = \{1,-2,-2,3,4\}$ [2+4]
- 3. Define Region of Convergence. Find inverse Z transform of $X(z) = z/\{(z-1)(z-2)^2\}$, ROC: |Z| < 1 [1+5]
- 4. Given H(z) for a system with the following difference equation:

y(n) = x(n) + x(n-2) [2+6+2]

Plot its poles and zeros in Z plane. Determine its magnitude response. Also, determine whether system is causal and stable.

- 5. Draw lattice structure for given pole zero system $H(z) = \frac{(0.5 + 2z^{-1} + 0.6z^{-2})}{(1 0.3z^{-1} + 0.4z^{-2})}$
- 6. What do you mean by Limit Cycle? How it occurs in recursive system? [1+3]
- 7. What is the condition satisfied by Linear phase FIR filter? Show that the filter with $h(n) = \{-1,0,1\}$ is a linear phase filter. [2+4]
- 8. Use Hanning window method to design a digital low-pass FIR filter with pass-band edge frequency (w_p) = 0.25π, stop-band edge frequency (w_s) = 0.35π where main lobe width of Hanning window is 8π/M, M is the filter length.
- 9. Why Spectral Transformation is required? [2]
- 10. Design a low pass digital filter by impulse invariance method to an approximate Butterworth filter, if passband edge frequency is 0.2π radians and maximum deviation of 0.5 dB below 0 dB gain in the passband. The maximum gain of -15 dB and frequency is 0.35 π radian in stopband, consider sampling frequency 1Hz.
- 11. Why do we need Discrete Fourier Transform (DFT) although we have Discrete-time Fourier Transform (DTFT)? Find circular convolution between $x[n] = \{1,2\} \text{ and } y[n] = u[n] u[n-4].$
- 12. How fast is FFT? Draw the butterfly diagram and compute the value of x(7) using 8 pt DIT-FFT for the following sequences: $x(n) = \{1,0,0,0,0,0,0,0,0,0\}$

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Examination Control Division 2069 Chaitra

Exam.		Regular	
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV / I	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing (CT704)

✓ Candidates are required to give their answers in their own words as far as practicable. ✓ Attempt All questions. ✓ The figures in the margin indicate Full Marks. ✓ Assume suitable data if necessary. 1. Define Energy and Power type signal with suitable example. Check the signal [2+2] $x[n]=\cos(2n\pi/5)+\sin(\pi n/3)$ is periodic or not. 2. Define LTI system. Find the output of LTI system having impulse response [1+4]h[n] = 2u[n] - 2u[n-4] and input signal $x[n] = (1/3)^n u[n]$. 3. State the properties of region of convergence (ROC)? Derive the time shifting property of [3+3]Z-transform. 4. Why do we need Difference Equation? Draw Pole-zero in Z-Plane and plot magnitude [2+2+6]response (not to the scale) of the system described by difference equation y[n]-0.4 y[n-1]+0.2y[n-2] = x[n]+0.1x[n-1]-0.06x[n-2]5. Determine the Direct Form II realization of the following [4] y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.252x(n-2)6. Compute the lattice coefficients and draw the lattice structure of following FIR system [6] $H(z) = 1 + 2z^{-1} - 3z^{-2} + 4z^{-3}$ 7. Design a digital FIR filter for the design of the low pass filter having [8] $\omega_p = 0.3\pi$, $\omega_s = 0.5\pi$, $\alpha_s = 40$ dB using suitable window function. 8. What is optimum filter? Describe Remez exchange algorithm for FIR filter design with [1+6] flow chart. 9. What is the advantage of bilinear transformation? Design a low pass discrete time Butterworth filter applying bilinear transformation having specifications as follows: Pass band frequency $(w_p) = 0.25\pi$ radians Stop band frequency $(w_s) = 0.55\pi$ radians Pass band ripple $(\delta_p) = 0.11$ And stop band ripple $(\delta_s) = 0.21$ Consider sampling frequency 0.5Hz Also, convert the obtained digital low-pass filter to high-pass filter with new pass band frequency $(w'_p) = 0.45\pi$ using digital domain transformation. 10. Why do we need FFT? Find 8-point DFT of sequence $x [n] = \{1,1,2,2,1,1,2,1\}$ using [2+7] Decimation in frequency FFT (DIFFT) algorithm. 11. Find $x_3[n]$ if DFT of $x_3[n]$ is given by $X_3(k) = X_1(k) X_2(k)$ where $X_1(k)$ and $X_2(k)$ are

4-point DFT of $x_1[n] = \{1,2,-2\}$ and $x_2[n] = \{1,2,3,-1\}$ respectively.

[6]

TRIBHUVAN UNIVERSITY

INSTITUTE OF ENGINEERING

Examination Control Division 2068 Bhadra

7-			2 . 3
Exam.	Reg	ular / Back	
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV / II	Time	3 hrs.

[9]

Subject: - Digital Signal Analysis and Processing

- Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt <u>All</u> questions.
- The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- 1. Find the energy and power of the signal x[n] = u[n].

[5]

- 2. Find the period of the signal $x[n] = \sum_{m=-\infty}^{\infty} \delta[n-2-3m]$. Find the Fourier series coefficients of the signal x[n]. [6]
- State whether or not the system $y[n] = e^{x[2n]}$ is (a) linear (b) time invariant (c) memoryless (d) causal. Where x[n] is input to system and y[n] is output of system. [5]
- 4. Convolve the sequences $x[n] = 3^n u[-n-5]$ and y[n] = u[n-5]. [5]
- 5. Find the frequency response of the linear time invariant system characterized by difference equation $y[n] = \frac{10}{24}y[n-1] + \frac{1}{24}y[n-2] = x[n]$. If input to the system is
 - $x[n] = \sin\left(\frac{\pi}{3}n\right) + \sin\left(\frac{\pi}{5}n\right)$ then determine output y[n] of the system. [7]
- Realize the overall system function:

 $H(z) = \frac{(1 - \frac{1}{5}e^{-j\frac{\pi}{5}}z^{-1})(1 - \frac{1}{3}z^{-1})(1 - \frac{1}{5}e^{j\frac{\pi}{5}}z^{-1})}{(1 - \frac{4}{5}z^{-1})(1 - \frac{1}{7}e^{j\frac{\pi}{7}}z^{-1})(1 - \frac{1}{5}z^{-1})(1 - \frac{1}{7}e^{-j\frac{\pi}{7}}z^{-1})}$

In terms of direct from I and direct from II structures. Draw the corresponding block diagrams of direct from I and direct from II structures.

How the spectrum of continuous time signal is related to spectrum of corresponding discrete time signal obtained by sampling the continuous time signal? Explain. Discuss what is aliasing and how it occurs.

[8] If passband edge frequency $\omega_p=0.25\pi$, stopband edge frequency $\omega_s=0.45\pi$, passband ripple $\delta_p = 0.17$ and stopband ripple $\delta_p = 0.27$ then design a digital lowpass Butterworth filter using bilinear transformation technique. [18]

- Use Blackman window method to design a digital low-pass FIR filter with passband edge frequency $\omega_p = 0.24\pi$, stopband edge frequency $\omega_s = 0.34\pi$ where main lobe width of Blackman window is $\frac{12\pi}{M}$, M is filter length. [9]
- 10. Use the Fast Fourier Transform decimation in frequency algorithm to find the discrete Fourier Transform of the sequence x[n] = [1 -2 2 1]. [8]

Examination Control Division 2063 Kartik

Exam.	Regular		
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV/II	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt <u>All</u> questions.
- ✓ <u>All</u> questions carry equal marks.
- ✓ Assume suitable data if necessary.
- 1. Explain the basic block diagram of discrete time processing of continuous signals. Write in detail about practical considerations during reconstruction.
- 2. Derive the expression for the discrete time Fourier series along with the power and energy expression in terms of Fourier coefficients.
- 3. Compute the DFT of the 4 points sequence x[n] = [3, 2, 1, 0] using the linear transformation.
- 4. Given the following two sequences x[n] = [11, 7, 0, -1], $-2 \le n \le 1$ and h[n] = [2, 3, 0, -5], $-1 \le n \le 2$, determine the convolution of x[n] and h[n] and plot it as well.
- 5. Given a sequence x[n] = [0, 1, 2, 3, 4, 5, 6, 7], determine X[k] using DIT or DIF FFT algorithm.
- 6. Given $\omega_p = 0.25\pi$, $\alpha_p = 0.5$ dB and $\omega_s = 0.55\pi$, $\alpha_s = 15$ dB, where the symbols have the usual meaning, find the order of filter and the cutoff frequency using Butterworth approximation.
- 7. Given $H[z] = \frac{1}{1 0.9z^{-1} + 0.64z^{-2} 0.576z^{-3}}$, find the 3rd stage lattice parameters for IIR systems and draw it as well.
- 8. Explain the steps of FIR filter design by using the Remez exchange algorithm.

Examination Control Division 2067 Mangsir

Exam.	Regular / Back		
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV / II	Time	3 hrs.

[2]

[4]

Subject: - Digital Signal Analysis and Processing

- Candidates are required to give their answers in their own words as far as practicable.
- Attempt All questions.
- The figures in the margin indicate Full Marks.
- ✓ Assume suitable data if necessary.
- Compute and plot even and odd component of the sequence x(n) = 2u[n] 2u[n 4]where u[n] is unit step sequence.
- 2. Write whether or not the following sequences are periodic and write the period.

a)
$$x[n] = \cos\left(\frac{5\pi}{3}n\right)$$

b)
$$x[n] = \sin\left(\frac{\pi n}{\sqrt{2}} + \frac{\pi}{8}\right)$$

- 3. Find the discrete Fourier coefficients of the periodic sequence with period N = 11 defined over a period as $x[n] = \begin{cases} 1, & |n| \le 2 \\ 0, & 2 < |n| \le 5 \end{cases}$ [4]
- 4. Show whether or not the system y(n) = nx[2(n-2)], n > 0 is (a) linear, (b) time invariant, (c) memoryless.
- [3] 5. Find the system function H(z) of the system characterised by difference equation $y[n] - \frac{5}{6}y[n-1] - \frac{1}{6}y[n-2] - x[n] = 0$. Find the poles and zeros of the system. Use the pole-zero diagram to plot the approximate frequency response magnitude of the system. [10]
- $H(z) = \frac{\left(1 \frac{1}{3}z^{-1}\right)\left(1 \frac{1}{4}z^{-1}\right)\left(1 \frac{1}{8}z^{-1}\right)}{\left(1 \frac{5}{6}z^{-1}\right)\left(1 \frac{1}{6}z^{-1}\right)\left(1 \frac{3}{4}e^{-j\frac{\pi}{4}}z^{-1}\right)\left(1 \frac{3}{4}e^{-j\frac{\pi}{4}}z^{-1}\right)} \text{ in }$ 6. Realize the system function H(z) = -

terms of cascade of second order sections. Draw the block diagram of the cascade realization.

- 7. Show by giving examples that the quantization error by truncation for sign magnitude number, e_{tsm} , lies in the range $-(2^{-b}-2^{-b_u}) \le e_{tsm} \le (2^{-b}-2^{-b_u})$ and that for the 2's complement number, e_{t2e} , lies in the range $-(2^{-b}-2^{-b}u) \le e_{t2e} \le 0$. b_u is the number of bits before quantization and b is the number of bits after quantization.
- 8. How does an IIR filter differ from an FIR filter?

- 9. Find the system function for digital filter using impulsive invariance technique from the analog Butterworth filter transfer function $H(s) = \frac{1}{(s+1.3)(s-1.3e^{\frac{j^2\pi}{3}})(s-1.3e^{-j\frac{2\pi}{3}})}$. T=1 second, and draw the block diagram of the system function, H(z), realized in terms
- 10. Show that the filter with impulse response h[n], $0 \le n \le N-1$, where h[n] = h[N-1-n], is a linear phase filter.

[15]

[6]

[5]

[8]

of second order sections.

- 11. Use the window method to design a digital low-pass FIR filter with Pass band frequency $(\omega_p) = 0.35\pi$, Stop band frequency $(\omega_s) = 0.45\pi$ with stop-band attenuation of at least 54dB.
- 12. Perform circular convolution of the sequences $x_1[n] = [1,2,1], 0 \le n \le 2$ and $x_2[n] = [1,2,0,1], 0 \le n \le 3$.
- 13. The duality property of Discrete Fourier Transform (DFT) is, if $x[n] \xrightarrow{DFT} X[k]$ then $X[n] \xrightarrow{DFT} nx[[-k]]_N$. For input sequence x[n] an algorithm can compute DFT using the formula $X[k] = \sum_{n=0}^{N-1} x[n] e^{-j\frac{2\pi}{N}kn}$. How can this same formula be used to find inverse discrete Fourier transform (IDFT) of input sequence as X[k] with output sequence as x[n] (use duality property)?

Exam.	Back		
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV / II	Time	3 hrs.

[6]

Subject: - Digital Signal Analysis and Processing.

Candidates are required to give their answers in their own words as far as practicable.

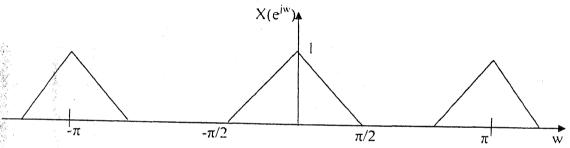
Attempt All questions.

The figures in the margin indicate Full Marks.

Assume suitable data if necessary.

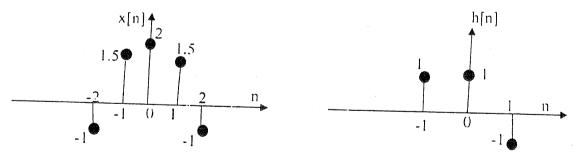
1. Define energy and power type signals with suitable examples. Check whether the unit step signal and unit impulse is energy type or power type with corresponding measure.

If $y[n] = x[n] \cos(\pi/2)n$; then draw $Y(e^{j\omega})$ for given $X(e^{j\omega})$.



2. Draw power density spectrum of given periodic signal $X(n) = \{1,0,1,1\}$ with fundamental periodic time N = 4.

[6] 3. Define recursive and non recursive system with suitable examples. Find the output of LTI system for input x[n] and impulse response h[n]. [2+5]



4. If $X_1[K]$ and $X_2[K]$ are DFT of sequence $X_1[n] = \{1,2,-1,3\}$ and $X_2[n] = \{0.5,1,2,1\}$ respectively, then find the sequence $X_3(n)$, if DFT of $X_3(n) = X_1(K) X_2(K)$. [6]

OR

Find X(4) and X(5) components of DFT values of sequence $x(n) = \{1,2,-1,3,2,0,1\}$ using any Radix-2 FFT algorithm.

5. Define steady state and transient response of an LTI system. Find the output of LTI system if input signal $x[n] = 15 - 7\sin(\pi/2)n + 20\cos n\pi$ and impulse of $h[n] = (1/2)^n u[n]$. [2+5] 6. The output of a system is characterized by y[n] = 0.2y[n-1] + 0.15y[n-2] + x[n] + $0.1 \times [n-1] - 0.72 \times [n-2]$. Show the pole zero of system, draw magnitude of response of system. (not to the scale)

[2+6]

7. What are the advantages of lattice structure over Direct Form Structure?

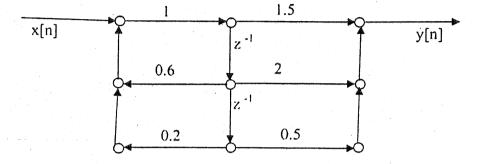
[4]

8. Show that the resolution of floating point representation is finer for smaller number and coarser for large number.

[4]

9. Draw lattice structure for given system in Direct Form II structure.

[8]



10. For an analog system function $H(s) = \frac{s+0.1}{(s+0.1)+9}$, convert it into digital IIR filter by

impulse invariance method and matched Z-transform method by selecting T = 0.1. Compare both digital T.F.

[8]

11. Describe Park McClellan algorithm with the help of type I linear phase FIR filter.

[8]

12. Write short notes on: (any two)

[4×2]

- a) Gibb's Phenomenon
- b) Kaiser window
- c) Characteristic of fixed window
- d) Limit cycle

Examination Control Division 2065 Baishakh

Exam.	Back		
Level	BE	Full Marks	80
Programme	BCT	Pass Marks	32
Year / Part	IV/II	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ Attempt <u>All</u> questions.
- ✓ The figures in the margin indicate <u>Full Marks</u>.
- ✓ Assume suitable data if necessary:
- 1. Define energy and power type signals with suitable equations. Calculate and plot Fourier coefficients for $x[n] = Sin (3\pi/5)n$ (4+5)
- 2. What is a stability? Explain it with suitable derivations and examples. (2+5)
- 3. If input sequences are x[-4] = 2, x[-2] = -1, x[0] = -2, and x[1] = 3, impulse responses to the system are h[-2] = 1, h[-1] = 0.75, h[0] = 0.5, and h[1] = 0.25. Calculate output sequences and plot input, impulse response, and output. (8)
- 4. Define a difference equation. Draw the block diagram for y[n] 2y[n-2] + 3y[n-3] 4y[n-4] = 3x[n] + x[n-1]. (2+5)
- 5. Differentiate between direct form I and II with suitable block diagrams. (7)
- 6. Compute and draw the lattice structure of given FIR filter. $H(z) = 2 + 0.35 z^{-1} + 0.3 z^{-2} + 0.45 z^{-3} + 0.55 z^{-4}$ (8)
- 7. Define One's complement, 2's complement and sign magnitude representation of numbers. Represent 80/136 in 8 bit 1's complement form. (6+2)
- 8. Explain about Park's McClellan algorithm with suitable derivation and flow chart.
- 9. Design a low pass discrete time filter by applying impulse invariance to an approximate Butterworth continuous filter, if passband frequency is 0.2π radians and maximum deviation of 1dB below 0dB gain in the passband. The maximum gain of -15dB and frequency is 0.3π radians in the stopband, Consider sampling frequency 1 Hz.
- 10. Find the FFT of the signal x[n] = (2, 1.5, 3.2, 1.7) (6)

Examination Control Division

2065 Magh

Exam.	Regular/Back		****
Level	BE	Full Marks	80
Programme	BCT (059 & Later Batch)	Pass Marks	32
Year / Part	IV/II	Time	3 hrs.

Subject: - Digital Signal Analysis and Processing

✓ Candidates are required to give their answers in their own words as far as practicable.

✓ Attempt <u>All</u> questions.

✓ <u>All</u> questions carry equal marks.

✓ Assume suitable data if necessary.

- 1. Explain in brief about the energy and power type signals. Determine the energy and power of the unit step sequence.
- 2. Explain recursive and non-recursive system. Show that a linear time invariant system is stable if its impulse response is absolutely summable.
- Compute DFT of the four point sequence.

 $x(n) = \{ 2 \ 3 \ 5 \ -2 \}$

4. Find the frequency response of the moving average filter. The impulse response of the moving average filter is

h(n)1/(M1+M2+1)otherwise

Sketch the magnitude and phase responses for $M_1 = 0$ and $M_2 = 4$.

- 5. Write down the general higher order difference equations. Find its system function. Draw the block diagram representations of Direct Form I and Direct Form II of the Nth order difference equations.
- 6. Explain in brief about bilinear transformation method of filter design. Also derive an expression for the continuous frequency (Ω) to the discrete time frequency (ω) as

 $\Omega = \frac{2}{T_d} \operatorname{Tan}\left(\frac{\omega}{2}\right)$, where T_d is the sampling interval.

- 7. What is Gibb's Phenomenon? Explain in brief about Kaiser Window filter design method.
- 8. Write short notes on:
 - a) Fast Fourier Transform (FFT)
 - b) Limit Cycles Oscillations