Candidates are required to give answers in their own words as far as practicable.
The figure in the margin indicates full marks. Assume suitable data if necessary.
Attempt all the questions.

No.		Question	Marks
1.	a)	What is digital Signal processing? Name the basic elements of digital signal processing and briefly explain each of them. List any two advantages of Digital signal processing.	8
	b)	Given a discrete time signal $x[n] = \{1, 1, 1, 1, \frac{1}{2}, \frac{1}{2}\}$, i. sketch $x[n]$, ii. Sketch $x[2-n]$ and iii. Determine $x[n]u[2-n]$	7
2.	a)	Verify the commutative property of convolution where input signal, x[n] = {1, 1, 1, 1} and response of system, h[n] = {1, 1, 1, 1}.	8
	b)	What are the two major characteristics of ROC? Determine the inverse z-transform of: $H(z) = \frac{1+2z^{-1}+z^{-2}}{1+4z^{-1}+4z^{-2}}$ (choose any one of the methods)	7
3.	a)	Define DFT and IDFT. Explain DFT as a linear transformation.	8
	b)	Find the circular convolution of the sequence: $x(n) = \{0,1,2,3\}$, $h(n) = \{2\ 1,1,2\}$	7
4.	a)	Determine the 4-point DFT of the following sequence using DIF FFT radix-2 algorithm: $x(n) = u(n) - u(n-4)$	8
	b)	Draw the lattice structure from the given FIR filter's system function: $H(z) = 1 + \frac{13}{12}z^{-1} + \frac{5}{8}z^{-2} + \frac{1}{3}z^{-3}$ **OR** Given a 3-stage lattice filter with coefficient $K_1 = \frac{1}{4}, K_2 = \frac{1}{2}$ and $K_3 = \frac{1}{3}$. Draw the corresponding FIR filter direct form I or II model.	7
5.	a)	Design a normalized linear phase FIR filter having the phase delay of 4 and attenuation of at least 40 dB in the stopband. Also obtain the magnitude response of the filter. **OR** Design a linear FIR phase filter using kaiser window to meet the following specifications: - $0.99 \le \|H(e^{jw})\| \le 1.01$ for $0 \le \|w\| \le 0.19\pi$	8

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

	$\lvert H(e^{jw}) \rvert \le 0.01, for\ 0.21\pi \le \lvert w \rvert \le \pi$ **b)** Design a lowpass FIR filter using frequency sampling technique having cut off frequency of $\frac{\pi}{2}$ rad/sample. The filter should have linear phase of 4 and length of 17.	7
6.	**a)** The transfer function of an analog lowpass filter is $H_a(s) = \frac{1}{s+1}$, and its bandwidth is 1 rad/sec. Design the digital IIR filter using Bilinear transformation method whose cut-off frequency is 0.2π and sampling interval is 0.0167 sec.	7
	b) Find the order and cut-off frequency of a digital filter by using impulse invariance method with the following specifications: $0.89 \le \lvert H(e^{jw}) \rvert \le 1,$ for $0 \le w \le 0.4\pi$ $\lvert H(e^{jw}) \rvert \le 0.18$, for $0.4\pi \le w \le \pi$ Also, plot its poles.	8
7.	**Write short notes on:** *(Any two)* **a)** Necessary and sufficient condition for a system to be stable **b)** Gibbs Phenomena in FIR filter design. **c)** LTI system	2×5

*** *Best of Luck* ***

A cos 2πft

Nepal College of Information Technology

Assessment

Semester- Fall

Level: Bachelor | Year: 2024

Programme: BE _CE | Full Marks: 100

Course: Digital Signal Analysis and Processing | Pass Marks: 45

Time: 3hrs.

Candidates are required to give their answers in their own words as far as practicable.
The figures in the margin indicate full marks.
Attempt all the questions.

1 a) A digital communication link carries binary coded words representing samples of an input signal $x_a(t) = 3\cos 600\pi t + 2\cos 800\pi t$. The link is operated at 10.000 bits/sec and each input sample is quantized into 512 different voltage levels. 8

i. What is the sampling frequency and folding frequency?
ii. What is the Nyquist rate for the signal $x_a(t)$.
iii. What are the frequencies in resulting discrete time signal x(n)?
iv. What is the resolution?

b) Define Energy and Power Signal. Determine the convolution sum of two sequences using graphical method where bold letter denotes origin. 7

x(n)={4,2,1,3} , h(n)={1, 2,2,1}

2 a) A causal LTI system is described by the difference equation 7

y(n)=y(n-1)+y(n-2)+x(n)+2x(n-1)

Find the system function and frequency response of the system. Plot the poles and zeros and indicate the ROC.

b) Find the Z transform and ROC of 8

$$x(n) = 2\left(\frac{5}{6}\right)^n u(-n-1)$$

3 a) Find the 8-point DITFFT of x(n)== $\sin\frac{3\pi n}{4}$ $for\ 0 \le n \le 7$ 8

b) Prove that multiplication of two sequences gives the result in circular of these two sequences. 7

4 a) Obtain the parallel form realization of following IIR filter. 7

$$H(z) = \frac{1+\frac{1}{4}z^{-1}}{(1+\frac{1}{2}z^{-1})(1+\frac{1}{2}z^{-1}+\frac{1}{4}z^{-2})}$$

b) Draw the lattice-ladder structure for the following IIR system. 8

$$H(z) = \frac{1+z^{-1}+2z^{-2}+z^{-3}}{1+\frac{13}{24}z^{-1}+\frac{5}{8}z^{-2}+\frac{1}{3}z^{-3}}$$

5 a) Determine H(z) using Impulse invariant technique for the analog system function 7

$$H_a(s) = \frac{1}{(s+1)(s^2+s+2)}$$

b) Design a low pass Butterworth digital Filter to give response of 3dB or less for frequencies upto 2kHz and attenuation of 20dB or more beyond 3kHz. Use the Bilinear transformation technique and obtain H(z) of the desired filter. Take sampling frequency as 8kHz. 8

6 a) Design a linear FIR filter using Kaiser window to meet the following specifications: 7

$$0.99 \le |He^{jw}| \le 1.01 \qquad 0 \le |w| \le 0.19\pi$$
$$|He^{jw}| \le 0.01 \qquad 0.21\pi \le |w| \le \pi$$

b) Design a filter with 8

$$H_d(e^{jw}) = \begin{cases} e^{-j3w} & -\frac{\pi}{4} \le w \le \frac{\pi}{4} \\ 0 & otherwise \end{cases}$$

Using a Hanning window with M=6

7 Write short notes on: **(Any two)** 5 x 2=10

a. Linear System
b. Properties of Fourier transform
c. Circular Convolution

Nepal Engineering College

Assessment (Fall Semester)

Level: Bachelor | Year: 2024

Program: Computer (5^{th} Sem) | Full Marks: 100

Course: Digital Signal Analysis &Processing | Time : 3hrs

- ✓ *Candidates are required to give their answers in their own words as far as practicable.*
- ✓ *Assume suitable data if necessary or missing.*
- ✓ *Attempt all the questions*

1 a Define Signal analysis and signal processing. Explain the advantages of digital signal processing over analog signal processing. 7

b Consider an LTI system with impulse response $h[n] = \{1, \underset{\uparrow}{4}, 2, -1\}$ and output 8

$y[n] = \{1, \underset{\uparrow}{4}, 13, 15, 4, -3\}$. Determine the input of the system, $x[n]$.

2 a Determine whether or not the given signal $y[n] = nx[n^2]$ is 7

i) Linear ii) Causal iii) stable iv) time-invariant v) with memory or not?

Or

Define elementary signals. Define any 5 elementary signals with mathematical function and neat diagrams.

b *Determine the z-transform and sketch the Roc of the following signal (use z-transform properties).* 8

$x[n] = na^n u[n-2]$.

3 a Draw the pole-zero plot and find the magnitude response of the following system: 7

$$H(z) = \frac{2}{1 - 0.5z^{-1}}$$

b The z-transform of the signal is described as 8

$$X(z) = \frac{2z^{-1} + 1}{3z^{-2} - 4z^{-1} + 1}$$

What are the possible ROCs? Determine the inverse z-transform for causal signal.

4 a Suppose we have two sequences x[n] and h[n] as follows: 7

$x[n] = \{1, 1, 4, -1, 7\}$; $h[n] = \{1, 4, 5, -2\}$.

Calculate circular convolution graphically.

b Find 8-point DFT of the sequence {0, 2, 5, 6, 0, -6, -5, -2} using radix-2 decimation in time domain. 8

5 a Consider the following system described by LCCD equation: 7
$y[n] = 0.6y[n-1] - 0.11y[n-2] + .006y[n-3] + x[n] + 2x[n-1]$.
Draw the direct form-II and cascade structures.

b Draw the lattice ladder structure for the following system: 8

$$H(z) = \frac{1 - 0.8z^{-1} + 2z^{-2}}{1 - 0.8z^{-1} - 0.86z^{-2}}$$

Is this system stable?

6 a Design a digital lowpass Butterworth filter to meet the following specifications using bilinear transformation method. 8
Passband cutoff frequency: $\omega_p = 0.15\pi$
Stopband cutoff frequency: $\omega_s = 0.55\pi$
Passband ripple: -3 dB.
Stopband ripple: -21 dB.
Sampling frequency = 3000 samples per second.

b Design linear phase FIR filter having the cutoff frequency of 0.6π, phase delay of $\tau = 4$ and atleast 48 dB attenuation in the stopband. 7

Or

How can you design FIR filter using windowing techniques? Explain with the help of fixed windowing technique.

7 Write short notes on (Any two): 5*2
a Computational complexity of DFT.
b Impulse invariance method of IIR filter design.
c Stability and causality of discrete time LTI system

Order of Filter N	Transfer function H(s) = 1/A(s) where A(s)
2	$(s^2+0.707s+1)$
3	$(s^2+s+1)(s+1)$
4	$(s^2+0.766s+1)(s^2+1.848s+1)$
5	$(s^2+0.618s+1)(s^2+1.618s+1)(s+1)$
6	$(s^2+0.518s+1)(s^2+4.414s+1)(s^2+1.932s+1)$
7	$(s^2+1.802s+1)(s^2+1.247s+1)(s^2+0.445s+1)(s+1)$

COSMOS COLLEGE
OF MANAGEMENT AND TECHNOLOGY

Term Test II

Date: 2081/11/01		Full Marks	100
Level	BE	Time	
Programme	BCE	3 hrs	
Semester	V		

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 a) *Test whether the given discrete-signal is period. If so, find the fundamental period.* 8

i) $x[n] = sin\left(\frac{2\pi}{3}\right)n$

ii) $x[n] = 4e^{j\frac{4\pi\left(n+\frac{1}{2}\right)}{5}}$

b) A digital discrete time signal $x[n] = 6.5\cos(0.1\pi n)$ is quantized with the resolution, $\Delta = 0.01$. How many bits are required in the A/D converter? If the maximum frequency that can be reconstructed from above signal is 500 Hz, determine the reconstructed time signal? 7

2 a) Find the response of a LTI system with input x(n) ={9, **8**, 7} to an impulse response h(n) = {5, -4, **8**}, where bold letter denotes origin. Use graphical approach. Also verify your answer. 8

b) Determine Z- transform and ROC of 7

$$x(n) = \left[\left(\frac{1}{2}\right)^n + \left(\frac{3}{4}\right)^n\right]u(n-5)$$

3 a) Find the 8-point DFT of the signal using decimation in time FFT. 8

$$x(n) = sin\frac{3\pi n}{4}\ ; for\ 0 \le n \le 7$$

b) Define zero padding. Find the circular convolution of $x_1(n) = \{1,2\ 3\}$ and $x_2(n) = \{9,8,5,1\}$. 7

4 a) Obtain the parallel form realization of the following IIR filter. 8

$$H(z) = \frac{3(2z^2 + 5z + 4)}{(2z+1)(z+2)}$$

b) Compute the lattice coefficient and draw lattice structure for given IIR system. Also check stability of the system. 7

$$H(z) = \frac{1}{(1-0.525z^{-1}+0.6125z^{-2}+0.3z^{-3})}$$

5 a) Design a lowpass filter which will have -3dB cut off at 30π rad/sec and an attenuation of 50 dB at 45π rad/sec. the filter is required to have a linear phase and the system uses a sampling rate of 100n samples/sec. 8

b) Design a linear FIR filter using Kaiser window to meet the following specifications: 7

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

6 a) Determine H(z) using impulse invariance method for the following analog filter's transfer function: 7

$$H_a(S) = \frac{1}{(S+0.5)(S^2+0.5S+1)}$$

b) Design a digital lowpass Butterworth filter using Bilinear Transformation method to meet the following specifications: 8

Passband attenuation = 1 dB

Stopband attenuation = 16 dB

Passband edge = 120 Hz

Stopband edge = 170 Hz

Sampling frequency = 1 KHz

7 Write short note: (Any two) 2x5

a) Symmetric and Anti-Symmetric filters

b) Gibbs phenomenon

c) Basic elements of DSP

National Academy of Science and Technology

Dhangadhi, Kailali

Pre-University Examination

Level: Bachelor Semester – Fall Year 2024

Programme: BE Full Marks : 100

Course: Digital Signal Analysis Processing Time : 3hrs.

Candidates are required to give their answers in their own words as far as practicable. The figures in the margin indicate full marks.

Attempt all the questions.

1. a) A digital communication link carries binary-coded words representing samples of an input signal $X_a(t) = 5\cos 300\pi t + 3\cos 900\pi t$. The link is operated at 10,000 bits/s and each input sample is quantized into 1024 different voltage levels. 8

   i) What is the sampling frequency and the folding frequency?

   ii) What is the Nyquist rate for the signal $X_a(t)$?

   iii) What are different frequencies in the resulting discrete-time signals x[n]?

   iv) What is the resolution Δ?

   b) Highlighting the key features of Digital Signal processing, briefly differentiate Digital Signal Processor with Analog Signal Processor. 7

2. a) Define LTI system. Mention and explain the properties of LTI system. 8

   OR

   Show that the multiplication of two DFT sequences results in circular convolution 7

   b) What do you understand by unit step response of an LTI system? Also find the unit step response of the system whose impulse response is given as h[n] = {1, 1, 2, 3, 1}.

   OR

   Obtain a linear convolution of the following two discrete-time signals:

   $x(n) = \{0,1,2,3\}$ and $h(n) = 2n[u(n)-u(n-3)]$

3. a) For the given system $y[n] = 0.7\,y[n-1] - 0.1\,y[n-2] + x[n]$, plot the pole zero diagram and magnitude response of the system 8

   b) Determine the causal signal x[n] having Z-Transform $X(z) = 1/\{(1-2z^{-1})(1-z^{-1})^2\}$. 7

4. a) Define DFT. Also compute 4-point DFT of the following sequence: 7

   $$x(n) = \{2, 0, 1, 2\}$$

   b) Find 8- point DFT of the sequence. {1,-1,2,3,1,1,2,3} using radix 2 DITFFT. 8

5. a) If there is given a three stage lattice filter with coefficient $K_1 = 1/4$, $K_2 = 1/2$, and $K_3 = 1/3$, determine the FIR filter coefficient for the direct-form structure. 8

b) Determine the lattice coefficients corresponding to the FIR filter with system function and also draw the lattice structure. 7

$$H(Z)=1+\frac{13}{24}z^{-1}+\frac{5}{8}z^{-2}+\frac{1}{3}z^{-3}.$$

6. a) Design an IIR filter by using Bi-linear transformation to meet the following specification $w_p = 0.25\pi$, $w_s = 0.55\pi$, $\alpha_s = 15\text{dB}$, $\alpha_p = 0.5\text{dB}$. 8

OR

Design a linear FIR filter using Kaiser window to meet the following

$$0.99 \le |H(e^{j\omega})| \le 1.01, for 0 \le |\omega| \le 0.19\pi$$

$$\le |H(e^{j\omega})| \le 0.01, for 0.21\pi \le |\omega| \le \pi$$

b) Draw the flowchart of Remez algorithm and explain it

7. Write short notes on (**Any Two**): 5×2

a) Gibb's Phenomena

b) Cascade structure for FIR system

c) ROC

Madan Bhandari College of Engineering

Urlabari-3, Morang

Final Internal Examination

Level: Bachelor of Computer Eng.	Full Marks: 100
Programme: BE	Pass Marks: 45
Year/Part: III/I	Time: 3 hrs

Subject: - **DSAP**

1. a) Explain the basic element of DSP system with a net block diagram. Discuss the at least there practical application of DSP system. (7)
   b) Explain the classification the properties of the signal. (8)

2. Determine whether the unit step signal is energy or power signal or neither. (5)
3. Find the output of LTI system having impulse response $h(n) = (½)^n [u(n+2)-u(n-2)]$ to the input $x(n) = [2,1,0,-1,4]$. (5)

4. a) State and prove the parseval's theorem of the z- transform and explain its physical significance. (7)
   b) Determine the z-transform and ROC of the signal $x(n) = \alpha^n u(n)$. (8)

5. Given the analog signal $X(t)= 3 \cos 2000\pi t +5\sin 6000\pi t+10\cos 12000\pi t$ and find the following. (10)
   a) What is the Nyquist rate? -
   b) What is the discrete time signal after samping , sampling rate is 5000 sample /sec.
   c) What is the data rate if each sample signal is quantised 1024 different voltage level?
   d) Find the resolution.
   e) What is the analog signal that can be reconstructed from samples.
6. Find the circular convolution of the sequence X1(n)=[1,2,3,4] and X2(n)=[-2,-1,2,3] in graphical method. (5)

7. a) Draw the direct-I and direct-II structure of following. (10)
   $Y[n]-0.75Y[n-1]-0.25Y[n-2] = X[n]+x(n-1)$
   b) Compute lattice and ladder coefficient.

8. Given x(n)={0,1,2,3}.find x(k) using DIT-FFT algorithm. (5)
9. Design a chebyshev low pass filter of 1 rad/sec bandwidth and following specification :
   a) Acceptable passband ripple of 2 DB.
   b) Cutt off frequency of 1 rad/sec
   c) Stopband attenuation of 20 dB or greater beyond 1.3 rad/sec. (10)

10. Design a second order discrete time butterworth filter with cut off frequency of 1khz and sampling frequency of 10^4 samples/sec by using bilinear transformation method. (10)

11. Design the linear phase FIR fiter using Kaiser window meet the following specification. (10)
    $0.99 \le |(He^{jw})| \le 1.01$ for $0 \le w \le 0.91\pi$
    $|(He^{jw})| \le 0.01$ for $0.21\pi \le w \le \pi$

United Technical College

Level: Bachelor
Programme: BE
Course: DSAP

Semester: Fall

Year:2024
Full Mark: 100
Pass Mark: 45

Candidates are required to give their answer in own words as far as practicable.
The figure in the margin indicates full marks.
Attempt all the questions.

1. a. Define DSP. Explain the applications of DSP in the field of communication engineering. [7]

   b. A Digital communication link carries binary coded words representing samples of an input signal
   $x_a(t) = 3cos\,?000\pi t + 2sin6000\pi t$ operated at 18,000 bits/s. and each input sample is quantized into 512 different voltage levels.
      i. What is the discrete time signal obtained after sampling?
      ii. What is the resolution Δ? [8]

2. a. A discrete time Signal x(n) is applied to a discrete time LTI system with unit impulse h(n) . Find output or response y(n) if given that
   $$x(n) = a^n u(n), 0 < a < 1 \text{ and } h(n) = u(n)$$ [7]

   b. Define Z transform. Determine the Z transform of
   $$x(n) = na^n u(n)$$ [8]

3. a. Find the inverse z - transform of the system
   $$X(z) = \frac{z+1}{(z+2)(z-1)(Z-1)}$$ [7]

   b. Obtain the circular convolution of the following sequences
   $$x_1(n) = \{1, 2, ?, ?\} and\ x_2(n) = \{4, 0, 2, .2\}$$ [8]

4. a. What is the importance of FFT? Find 8-point DFT of sequence x(n) = {1, 2, 0, 5, 3} using DIT FFT algorithm [7

   b. Obtain cascade and parallel structure for the following system

$y(n) = y(n-1) - 0.5y(n-2) + x(n) - x(n-1) + x(n-2)$ [8]

5. a. Draw lattice and ladder structure [7]

$$H(z) = \frac{1 - z^{-1} + 0.5z^{-2}}{1 + 0.2z^{-1} - 0.15z^{-2}}$$

b. Design an FIR filter to meet the following specifications [8]

Passband = 2 kHz
Stop band = 5 kHz
Stop band attenuation = 42 dB
Sampling frequency = F_s = 20 kHz
Use Hamming Window

6. a. Determine H(z) using impulse invariance method for the following system function

$$H_a(s) = \frac{s+0.1}{((s+0.1))^2 + 3^2}$$ [7]

b. Using bilinear transformation design a Butterworth filter which satisfies the following conditions [8]

$0.8 \leq |H(e^{j\omega})| \leq 1 \qquad 0 \leq w \leq 0.2\pi$

$|H(e^{j\omega})| \leq 0.2 \qquad 0.6\pi \leq \omega \leq \pi$

7. Write shorts note on **(any two)**
   a. Computational Complexity of FFT
   b. Recursive and non-recursive system
   c. Remez Exchange Algorithm

LUMBINI ENGINEERING MANAGEMENT AND SCIENCE COLLEGE
Bhalwari, Tilottama

Level: Bachelor
Programme: BE
Course: Digital Signal Analysis and Processing (Comp.5th Semester)

Year: 2025
Full Marks: 100
Pass Marks: 45
Time: 3 hrs.

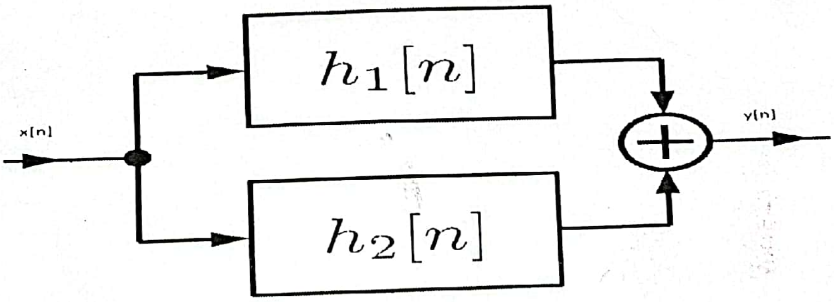
Candidates are required to give their answers in their own words as far as practicable.
The figure in the margin indicates full marks.
Attempt all the questions.

1.

a. A digital communication link carries binary code words representing samples of an input signal; $x_a(t) = 5\cos 600\pi t + 7\cos 800\pi t$. The link is operated at 1000 bits/sec and each input sample is quantized into 1024 different voltage levels. [2+1+2+2]

i. What is the sampling frequency and folding frequency?
ii. What is the Nyquist rate for the signal $x_a(t)$?
iii. What are the frequencies in the resulting discrete time signal x[n]?
iv. What is the resolution 'Δ'?

b. Two subsystems $h_1[n]$ and $h_2[n]$ are interconnected as shown in the block diagram. Determine the response of the system if; $h_1[n] = \{1,4,2\}$ and $h_2[n] = \{2, 3\ 1\}$, when excited by input; $x[n] = \{2, 4\}$. [8]



2.

a. Prove that a discrete time LTI system is stable if and only if it's impulse response is absolutely summable. Determine whether the given discrete time system described by LCCDequation is [5+2]

i. Time invariance

$$y[n] = -2y^2[n-1] + 3x[n] + 2x[n-1]$$

[8]

b. Determine the inverse Z-transform of

$X(Z) = \frac{1}{1-1.5z^{-1}+0.5z^{-2}}$; when the ROC is

i. ROC: $|z| > 1$
ii. ROC: $|z| < 0.5$
iii. ROC: $0.5 < |z| < 1$

Also specify the causality and stability in each case.

[OR]

State the condition for the stability for the Z-transformed of sequence x[n]. Also state and prove the convolution property of Z-transform

3

a. Compute the 4-point DFT of the sequence

$X_a(t) = 4\cos 200\pi t$, with sampling frequency of 500 Hz. 7

b. Determine the response of the system using FFT algorithm, if the input x[n]and impulse response h[n] are given as under; 8

x[n]={2, 2, 4} and h[n]={1, 1}

a. Determine the cascade and parallel realization of the discrete time system described by differential equation. [8]

$$y[n] = -\frac{3}{4}\, y[n-1] + \frac{1}{4}\, y[n-2] + x[n] + \frac{1}{2}\, x[n-1]$$

b. Obtain the lattice ladder structure of the discrete time system described by the differential equation [7]

$$y[n] = -\frac{3}{4}\, y[n-1] + \frac{1}{4}\, y[n-2] + x[n] + \frac{1}{2}\, x[n-1]$$

Also check the stability of the filter

5.

a. Design a digital low pass Butterworth filter by applying bilinear transformation technique for the given specifications. [8]

Pass band edge = 120Hz

Pass band attenuation = 1dB

Stop band edge = 170Hz

Stop band attenuation = 16 dB Assume

sampling frequency of 512 Hz

b. Obtain H(z) using the impulse invariant techniques for an analog system function which is given by: [7]

$$H_a(s) = \frac{1}{(s+0.5)(s^2+0.5\,s+2)}$$

6.

a. Design a low pass digital filter to be used in A/D and D/A structure that will have -3 dB cut-off at $30\pi\ rad/sec$ and an attenuation of 50 dB at $45\,\pi \frac{rad}{sec}$, the filter is required to have a linear phasse and the system uses sampling rate of 100 samples/second. [7]

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

$$0.99 \le \left|H(e^{i\omega})\right| \le 1.01, \quad \text{for } 0 \le |\omega| \le 0.19\pi,$$

$$\left|H(e^{i\omega})\right| \le 0.01, \quad \text{for } 0.21\pi \le |\omega| \le \pi$$

7. Write short note on(Any two) 2*5=10

a. Recursive and non-recursive system

b. FIR Vs IIR

c. Circular Convolution Vs linear convolution

PokharaEngineering College
Internal assessment- 2025

Level: Bachelor
Programme: BE
Course: Digital Signal Analysis and Processing

Year: 2025
Full Marks: 100
Time: 3 hrs.

Candidates are required to give their answer in their own words as far as practicable.
Figures in the margin indicate full marks.
Attempt all the questions.

1. a) Define signal processing? Compare between energy signal and power signal. 8

b) Explain the basic elements of a digital signal processing system. 7

2. a) A first order IIR system defined by the difference equation 7

$$y[n] - ay[n-1] = x[n]$$

Find:

i. System function
ii. Condition for stability
iii. Impulse response

b) Plot the magnitude and phase response of the system which has pole pair at $r = 0.9$ and $\theta = \frac{\pi}{4}$. 8

3. a) Determine the causal signal x(n) having the Z-transform 7

$$X(Z) = 1/[(1-2z^{-1})(1-z^{-1})]$$

b) Define ROC. Explain the properties of ROC. 8

4. a) Design a lowpass filter to be used in an ADC-H(z)-DAC structure that will have a $-3\ dB$ cutoff of $30\pi\ rad/sec$ and an attenuation of 40 dB at $45\pi\ rad/sec$. The filter is required to have a linear phase. Use a sampling rate of 100 $samples/sec$. 7

b) Design an FIR linear phase FIR filter using Kaiser window to meet the following specifications.

$$0.99 \le |H(e^{j\omega})| \le 1.01, \qquad 0 \le |\omega| \le 0.4\pi$$
$$0 \le |H(e^{j\omega})| \le 0.001, \qquad 0.6\pi \le |\omega| \le \pi$$

3+5

5. a) Obtain the cascade structure of the system with difference equation

$$y[n] = 3/4\ y[n-1] - 1/8\ y[n-2] + x[n] + 1/3\ x[n-1]$$ 7

b) Given a three-stage lattice filter with coefficients $K_1 = 1/4$, $K_2 = 1/4$, $K_3 = 1/3$, determine the FIR filter coefficients for the direct-form structure. 8

6. a) Design a low pass discrete- time filter by applying impulse invariance to an approximate Butterworth continuous filter, if passband frequency is 0.5π radians and maximum deviation of 1dB below 0 dB gain in the passband. The maximum gain of -40 dB and frequency is 0.7π radians in the stopband. 8

b) Discuss the FIR filter design by BLT method. 7

7. Write short notes on the following (any two): 2×5

a) Convolution sum

b) Elementary discrete-time signals

c) Gibb's phenomenon

Pokhara University

Everest Engineering College

Final Internal Assessment

Fall-2024

Level: Bachelor

Program: BE CMP (5th Semester)

Faculty: Science & Technology

Subject: Digital Signal Analysis & Processing

F.M. 100

P.M. 45

Time: 3hrs

Attempt all the questions.

1 A) What do you mean by signal processing? Differentiate between analog signal processing and digital signal processing. 7

B) Obtain the convolution sum of given two sequences h[n]= {1,2,1,2} and h[n]= {2,1,2,1}. Here bold elements denote the origin. Also, specify the value at origin in output. 8

2 a) Define Z-transform and Region of convergence (ROC). Find the Z-transform of the given sequence: 7

$$x(n) = r^n \sin\omega n \, u(n)$$

b) Find the solution to the difference equation 8

$$y(n) - \frac{3}{2}y(n-1) + \frac{1}{2}y(n-2) = \left(\frac{1}{4}\right)^n$$

For n≥0 with initial conditions: y (-1) =4 and y (-2) = 10

3 a) Obtain the direct Forms I and II realizations for a third order IIR transfer function which is expressed as below: 8

$$H(z) = \frac{0.56z^{-1} + 0.319z + 0.04}{0.5z^3 + 0.3z^2 + 0.17z - 0.2}$$

b) Determine the lattice coefficients corresponding to the FIR filter with system function 7

$$H(z) = A_3(z) = 1 + (13/24)\, z^{-1} + (5/8)\, z^{-2} + (1/3)\, z^{-3}.$$

4 a) Define Frequency wrapping. Differentiate between Impulse Invariance Method and Bilinear Transformation Method for IIR filter design. **7**

b) Using bilinear transformation, design a Butterworth filter which satisfies the following conditions: **8**

$0.89 \le |H(e^{j\omega})| \le 1$ for $0 \le \omega \le 0.4\pi$

$|H(e^{j\omega})| \le 0.18$ for $0.6\pi \le \omega \le \pi$

Use Bilinear Transformation for T = 1 Sec.

5 a) How can you design FIR filter using rectangular window? Explain. **7**

b) Explain the steps of designing FIR filters using Kaiser Window. **8**

6 a) Perform the 8-point FFT of the following sequence using Radix-2 algorithm using Decimation in Frequency (DIF). **8**

$x[n] = n + 1$ for $0 \le n \le 7$.

b) What is zero padding? Perform the circular convolution of the following two sequences using matrix method. **7**

$x_1[n] = \{1,2\}$ and $x_2[n] = u[n] - u[n-4]$.

7 Write short notes on: (Any two) **(2*5=10)**

a) Discrete time system properties.

b) Gibb's Phenomena

c) Energy and Power Signals

Good Luck