Dsap QNset - its dsap qn set for pou engineering students

Digital signal analysis and processing (Pokhara University)

POKHARA UNIVERSITY

Level: Bachelor | Semester: Spring | Year : 2023
Programme: BE | 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.

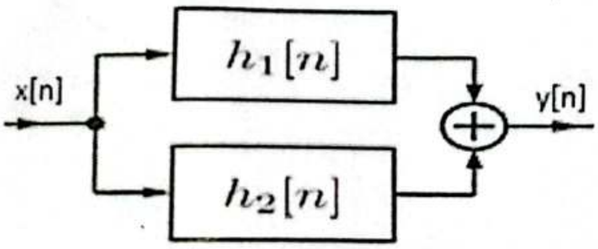
The figures in the margin indicate full marks.

Attempt all the questions.

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

   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 aliasing effect?

   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) Determine which signals are periodic and compute fundamental period. 8

   i. $\text{Cos}\left(\frac{\pi n^2}{8}\right)$ ii. $\text{Sin}\left(\frac{3\pi}{2}\right)\ \text{Cos}\left(\frac{5\pi n}{4}\right)$

   b) Define z-transform and Region of Convergence. Find z-transform of following signal $x[n] = a^n u[n]$. 7

3. a) Why do we need DFT when we have DTFT? Determine the circular convolution of the sequence: $x_1(n) = \{4,3,2,1\}$ and $x_2(n) = \{8,7,6,5\}$ 7

   b) Determine the direct form realization of the following difference equation. 8

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

Compare FIR and IIR

4. a) Determine the cascade and parallel realization of the discrete time systemdescribed by differential equation. 8

$$y[n] = -\frac{3}{4}\, y[n-1] + ¼\; y[n-2] + x[n] + \frac{1}{2}\, x[n-1]$$

b) Obtain the lattice ladder structure of the discrete time system described bythe differential equation 7

$$y[n] = -\frac{3}{4}\, y[n-1] + ¼\; y[n-2] + x[n] + \frac{1}{2}\, x[n-1]$$

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. 7

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: 8

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

6. a) Design a linear FIR filter using Kaiser window to meet the following specifications: 8

$0.99 \leq |H(e^{\wedge}jw| \leq 1.01$; for $0 \leq |w| \leq 0.19\pi$

$|H(e^{\wedge}jw| \leq 0.01$; for $0.21\pi \leq |w| \leq \pi$

b) By using Hanning window, design a low pass filter to approximate the ideal response given by: 7

$$H(e^{jw}) = \begin{Bmatrix} 1 & for & -\frac{\pi}{6} \leq w \leq \frac{\pi}{6} \\ 0 & & otherwise \end{Bmatrix}$$

Use the filter length of M=9 for your design

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

a) Sampling and quantization of analog signal,

b) BIBO Stable

c) Frequency Sampling

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Attempt all the questions.

a) Consider the analog signal $x_a(t) = 3\cos(100\pi t)$ 8

i) Determine the minimum sampling rate required to avoid aliasing.

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

iii) If the signal is sampled at the rate $F_s = 200$Hz. What is the discrete-time signal obtain after sampling?

iv) What is the frequency $0 < F < F_s/2$ of a sinusoid that yields samples identical to those obtained for F_s=200Hz

b) Define time-invariance and causality of a discrete time system. Examine whether following systems are stable or not: 7 2

i) $y[n] = x[n] + 1$

ii) $y[n] - y[n-1] = x[n] + x[n-1]$

a) Find the convolution between two signals $x[n] = a^n, for\ 0 \le n \le 6$ and $h[n] = 1, for\ 0 \le n \le 4$ 8 2

b) Define Z transform. Find the inverse z transform of $X(z) = \log(1 + az^{-1})$ 7 3

a) Why do we need DFT when we have DTFT? Determine the circular convolution of the sequence 8 4

$$x_1(n) = \{1, 2, 3, 4\}\ and\ x_2(n) = \{5, 6, 7, 8\}$$

Also verify your answer.

b) Define Fast Fourier Transform (FFT)? Determine 8-point DFT of the sequence using Decimation in Time FFT. 7 4.5

$$x(n) = cos\left(\frac{n\Pi}{4}\right)$$

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

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

5 b) Convert the following IIR filter into lattice ladder structure 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) Convert the following in Direct form- II realization described by 7

$H[z] = (1+z^{-1}+2z^{-2}+3z^{-3})/(1+15z^{-1}+15z^{-2}+12z^{-3})$

6 b) With the low pass specification Ω_p = 3.2 KHz and Ω_s = 4.8 KHz and sampling frequency Ωfs = 12 KHz and α_s = 40 dB, find the length and value of β for Kaiser Window. 8

7.2 6. a) Derive Impulse Invariant Method for IIR filter design. Also illustrate the mapping from the s-plane to the z-plane while using IIM. 7

7 b) Design a Chebyshev analog filter with maximum passband attenuation of 2.5 dB at $\Omega_p = 20\ rad/sec$ and stopband attenuation of 30 dB at $\Omega_s = 50\ rad/sec$ 8

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

a) Gibbs Phenomena 6.1

b) Basis Elements of DSP 1

2 c) BIBO Stable

POKHARA UNIVERSITY

Level: Bachelor	Semester: Spring	Year : 2021
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Course: Digital Signal Analysis and Processing		Pass Marks: 45
		Time : 3 hrs.

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

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Attempt all the questions.

1. a) Define energy and power signal with examples. Check whether the given signal is energy or power signal x(t)=Asin(t). 8 1

   b) Prove that an LTI system is causal if the impulse response of the system is a causal signal. 7 2

2. a) Find the convolution between two signals $x[n] = a^n, for\ 0 \le n \le 6$ and $h[n] = 1, for\ 0 \le n \le 4$ 8 2

   b) Define Z-transform. Find Z-transform of signal x[n]= $(\frac{1}{2})^n u[n] + (1+j)^n u[-n-1]$ and indicate ROC graphically. 7 3

3. a) Why DFT is needed and how the DFT solve the problem associated with DTFT. Use DIF-FFT algorithm to compute 8 point DFT of x[n]={2,1,1,1}. Discuss the result. 8 4

   b) Show that the multiplication of two DFT sequences results in circular convolution. 7 4

4. a) Realize the transfer function using cascade realization 7 5

$$X(Z)=\frac{2(z+2)}{Z(z-0.1)(Z+0.5)(z-0.4)}$$

   b) Draw the lattice structure of the given transfer function 8 5

$$H(Z)= 1+\frac{3}{8}z^{-1}+\frac{5}{4}z^{-2}+\frac{3}{2}z^{-3}$$

5. a) Design a FIR system to meet the following specifications 8 6

   Pass band edge frequency=2KHz

   Stop band edge frequency=5KHz

   Stop band attenuation=42dB

   Sampling frequency=20KHz

7 b) Derive Impulse Invariant Method for IIR filter design. Also illustrate the mapping from the s-plane to the z-plane while using IIM.

6. 7 a) Design a second order discrete-time Butterworth filter with cut-off frequency of 1 KHz and sampling frequency of 10^4 samples/sec by using bilinear transformations. Also plot the poles of the filter.

6'3 b) Design a linear FIR filter using Kaiser window to meet the following specifications:

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

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

7. Write short notes on: **(Any two)**

a) Region of Convergence 3.?

b) Causal and Non-Causal system 2

c) Gibbs Phenomena 6.1

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| | | Time : 3hrs.

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Attempt all the questions.

a) Define Signal Analysis and Signal Processing. What are the advantages of Digital signal processing over Analog signal processing? 8 1

b) The output of an LTI system is y (n) = {2, 5, 1, <u>-10</u>, -10, -3, 6, 9} to an impulse response h(n) = {1, <u>2</u>, 0, -3}. Find the input x(n) using convolution sum. The underline represents the element at n = 0. 7 2.5

a) Find the convolution between two signals 7 2

$x[n] = u[n]$ and $h[n] = a^n u[n], \ 0 < a < 1$

b) Determine Z–transform along with ROC of 8 3

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

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

b) What is Zero Padding ? Find the circular Convolution of $x_1[n] = \{1,2\}$ & $x_2[n] = \{3,2,1\}$ 7 4

4. a) Obtain the parallel form realization of following IIR filter. 7 5

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

b) Compute Lattice coefficients and draw lattice structure for given IIR system. Also check stability of the system. 8 5

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

5. a) Design a lowpass filter which will have -3 dB 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 100 samples/sec. 8 6.4

b) Design a linear FIR filter using Kaiser window to meet the following specifications: 6.3 7

$$0.99 \leq |H(e^{j\omega})| \leq 1.01\,, \quad for \;\; 0 \leq |\omega| \leq 0.19\pi$$
$$\leq |H(e^{j\omega})| \leq 0.01\,, \quad for \;\; 0.21\pi \leq |\omega| \leq \pi$$

7.2

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

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

7.1

b) Design a digital lowpass Butterworth filter using Bilinear Transformation method to meet the following specifications:
Passband attenuation = 1 dB
Stopband attenuation = 16 dB
Passband edge = 120 Hz
Stopband edge = 170 Hz
Sampling frequency = 256 Hz

7. Write short notes on: (**Any two**)
   a) Region of Convergence 3.?
   b) Symmetric and Anti-symmetric filter ?
   c) Kaiser window 6.3

POKHARA UNIVERSITY

Level: Bachelor	Semester: Fall	Year : 2020
Programme: BE		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.
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Attempt all the questions.

1. a) Define Signal Analysis and Signal Processing. What are the advantages of Digital signal processing over Analog signal processing? 8 1

   b) Define elementary signals. State and prove Convolution and time shifting property of Discrete time Fourier transform. 7 2

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 2

   b) Define Z–transform and ROC. Determine the Z-transform and plot the ROC for the given signal 7 3

$$x(n) = (-1)^n Cos\left(\frac{\pi}{4} n\right) u(n)$$

3. a) Why we need DFT when we have DTFT? Find the circular convolution of the sequence $x_1(n)$={9,8,7,6} and $x_2(n)$={6,4,3,2} using convolution sum. 8 4

   b) How efficient is FFT? Determine 8-point DFT of the sequence x(n) = {1, 0, 2, 0, 3, 0, 1, 1} using Decimation in Frequency Radix-2 Butterfly structure (DIFFFT). 7 4

4. a) Determine the direct form realizations of the following difference equation. 8 5

$$2y(n) + 3y(n-1) + 5y(n-2) = x(n) + 2x(n-3)$$

   Compare FIR and IIR.

   b) Compute the lattice coefficients and draw the lattice structure of following FIR system 7 5

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

5. 6.1 a) How can you design FIR filter using rectangular window? Explain.

6 b) Design a lowpass FIR with 7 coefficients for the following specifications.

Passband Frequency edge = 300 Hz

Sampling Frequency = 1 KHz

Use Hanning window for your design.

7.1, 7.3 6. a) Compare and Contrast Impulse Invariance Method and Bilinear Transformation Method of designing IIR filter.

7.1 b) Design a lowpass IIR filter to meet the following specifications

- Passband attenuation : 1 dB
- Passband frequency: 1.2 KHz
- Stopband attenuation: 40 dB
- Stopband frequency: 2.5 KHz
- Sampling Frequency: 10 KHz

7. Write short notes on: **(Any two)**

a) Kaiser window 6.3

b) Causality and stability of LTI system 2.

c) Sampling of CT signal

POKHARA UNIVERSITY

Level: Bachelor	Semester: Spring	Year : 2019
Programme: BE		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.

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Attempt all the questions.

1. a) Consider an analog signal: 8 1

$$x_a(t) = 3\cos 100\pi t + 2\sin^2 180\pi t$$

i) What is Nyquist rate for this signal?

ii) Assume that the samples of a given signal is operated at 2100 bits/s and each input sample is quantized into 128 different voltage levels. What is the discrete-time signal obtained?

iii) Determine the time period of the discrete-time signal.

iv) What is the resolution Δ?

b) Illustrate the significance of convolution summation in digital signal analysis. Compute and plot the convolution of $x[n] = \{0, \underset{\uparrow}{1}, 5, 4, 0\}$ and $h[n] = \{5, 3, \underset{\uparrow}{2}, 1, 0\}$. 7 2

2. a) Resolve the discrete time signal into impulses. Derive the equation for convolution sum for discrete-time LTI system. 8 2

b) Find the Z-transform of $x[n] = a^n\cos(w_0 n)u[n]$. Also specify ROC. 7 3

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

b) Show that the multiplication of DFTs of two sequences result in the circular convolution in time domain. 7 4

4. a) Obtain the parallel form realization of following IIR filter 5 7

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

b) A LPF has the desired frequency response 6 8

$$H_d(e^{jw}) = \begin{cases} e^{-j3\omega}\ ;\ 0 \le |\omega| < \frac{\pi}{2} \\ 0\ ;\ \frac{x}{2} < |\omega| \le \pi \end{cases}$$

6 Determine the filter coefficients $h(n)$ for N=7 using type –I frequency sampling technique.

5. 6.3 a) Show that the Kaiser window includes the rectangular window as a special case.

6.2 b) Use Hanning window method to design low-pass FIR filter with pass-band edge frequency $(\omega_p) = 0.4\pi$ & $M = 11$. Where M is the filter length.

7.1 6. a) Design a digital lowpass Butterworth filter using Bilinear transformation method to meet the following specifications:

Passband attenuation 3 dB	Passband edge = 2 KHz
Stopband attenuation 30 dB	Stopband edge = 6 KHz

Use sampling frequency = 10 KHz.

7.2 b) Determine H(z) using impulse invariance method for the following analog filter's transfer function :

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

7. Write short notes on: **(Any two)**

a) Energy Signal Vs Power Signal 1

b) Computational Complexity of DFT 4

c) Gibbs Phenomena 6.1

POKHARA UNIVERSITY

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Course: Digital Signal Analysis and Processing		Pass Marks: 45
		Time : 3hrs.

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Attempt all the questions.

1. a) Test whether the given discrete-signal is periodic. If so, find the fundamental period. 7 1

   i. $x[n] = \sin\left(\frac{2\pi}{3}\right)n$ ii. $x[n] = 4e^{j\frac{4\pi\left(n+\frac{1}{3}\right)}{5}}$

   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 500Hz, determine the reconstructed time signal? 8

2. a) State and prove the necessary and sufficient conditions for an LTI system to be causal and stable. 7 2

   b) Consider an LTI system with impulse response $h[n] = u[n] - u[n-5]$ and input $x[n] = (3/5)^n\{u[n]-u[n-4]\}$. Determine the output of the system, $y[n]$. 8 2

3. a) Determine the causal signal $x[n]$ if its Z-transform $X(z)$ is given by 7 3

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

   b) With the help of N = 8, explain radix-2 decimation-in-time (DIT) FFT algorithm for computation of DFT. Give the computational efficiency of FFT over DFT. 8 4

4. a) Using circular convolution method, determine the linear convolution of the following sequences: $x[n] = \{\underset{\uparrow}{1}, 2, 4\}$ and $h[n]$ = 8 4

   $h[n] = \{\underset{\uparrow}{1}, 2, 1, 3\}$.

   b) What are the advantages of representing the digital filter in the block 7 5 diagram form? Draw the direct form structures of the system described by LCCD equation

$y[n] = 0.3y[n-1] - 0.9y[n-3] + x[n] + 2x[n-1] + 4x[n-2]$

5. 7 a) Design a digital Butterworth low pass filter whose transfer function is given by

$$\begin{cases} -3.098dB \le |H(e^{jw})| \le 0 & 0 \le w \le 0.2\pi \\ |H(e^{jw})| \le -10.46dB & 0.6\pi \le w \le \pi \end{cases}$$

Use impulse-invariant transformation.

5 b) Draw the lattice structure for the following IIR system.

$$H(z) = \frac{1}{3+6z^{-1}+9z^{-2}}$$

Is the system stable?

6. a) Define symmetric and Anti-symmetric filter, and discuss the applications.

7 b) Design a lowpass filter which will have -3 dB cut-off at 30π rad/sec and an attenuation of 55 dB at 48π rad/sec. The filter is required to have a linear phase and the system uses a sampling rate of 200 samples/sec.

7. Write short notes on: 2

a) Filter Design by Kaiser Window

b) Compare and constrast FIR and IIR (imp)

c) Frequency shift property of DFT

POKHARA UNIVERSITY

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Course: Digital Signal and Analysis and Processing | Pass Marks: 45
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Attempt all the questions.

1. a) Define Signal, System and Signal Processing. Explain the limitations of DSP. 7 1

   b) What is LTI system? Obtain the necessary and sufficient condition for causality of LTI system. 8 2

2. a) State and prove time shifting property of DTFS. 7 4

   b) Define z-transform. Determine the z-transform and ROC of the signal x[n] = nan u[n] . 8 3

3. a) Define DFT. Determine the FFT of the signal 8 4

   x[n] = u[n] – u[n-4]

   using DIT-FFT algorithm.

   b) Determine the Circular convolution of the signals 7 3

   X1 [n] = { 1,2,2,1} and x2 [n] = {2,1,1,2}

4. a) A certain discrete-time filter has the following data: 7 5

   Poles are at 0.2 and 0.4.

   Zeros are at -0.4 and origin.

   Gain of filter is 5.

   Determine cascade form realization

   b) A system has an impulse response 8 5

   h[n]= (0.5)nu[n] + n(0.2)n u[n]

   Determine parallel form realization.

5. a) Design a normalized linear phase FIR filter having the phase delay of τ = 4 and at least 40 dB attenuation in the stop band. Also, obtain the magnitude /frequency response of the filter. 8 6

   b) Explain the FIR filter design by Kaiser Window. 7 6'3

6. a) If $H(s) = 1/[(s+1)(s+2)]$, find the corresponding H(z) using IIM method for sampling frequency of 5 samples/sec. 7

   b) Explain the design of IIR filter using IIM method. Also write the limitations of IIM method. 8 7

7. Write short notes on: (**Any two**)

a) Sampling of analog signals
b) Recursive and non-recursive systems
c) Radix-2 FFT algorithm

POKHARA UNIVERSITY

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Attempt all the questions.

1. a) What are the basic elements of a digital signal processing system? Explain each elements briefly 7 1

   b) Determine the response of the relaxed system characterized by the impulse response $h[n] = 0.5^n u[n]$ and input $x[n] = 3^n u[n-1]$. 8 2

2. a) Explain about the Time shifting and Time reversal properties of Discrete Time Fourier Transform. 7 4

   b) Write down the properties of ROC of Z-transform with examples. 8 3

3. a) Find the DFT of the sequence 7 4

   $x(n) = \cos\frac{\pi}{2} n \; ; for \; 0 \le n \le 7$

   Using radix–2 Decimation in Time Fast Fourier Transform (DITFFT) algorithm and keep track of all the intermediate quantities by putting them on the diagrams.

   b) Prove that multiplication of two sequences gives the result in circular of these two sequences. 8 4

4. a) Draw the lattice-ladder structure for the following IIR system. 7 5

$$H(z) = \frac{1 - 0.8z^{-1} - 0.15z^{-2}}{1 + 0.1z^{-1} - 0.72z^{-2}}$$

   Is the system stable?

   b) Obtain the direct form I and direct form II realizations for the system described by the following equation: 8 5

   $y(n) - 3y(n-3) = x(n) + 4x(n-3)$

5. a) Define Digital Filters. Differentiate between IIR & FIR Digital Filters. 8 6

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

$$0.98 \le |H(e^{j\omega})| \le 1.02, \quad for\ 0 \le |\omega| \le 0.18\pi$$
$$|H(e^{j\omega})| \le 0.02, \quad for\ 0.22\pi \le |\omega| \le \pi$$

6 6. a) Design an FIR filter with 5 coefficients for the following specifications:

- Passband edge frequency = 0.25 KHz
- Sampling frequency = 1 KHz
- Stopband attenuation = 40 dB

1 b) Convert the analog filter with system function

$$H(s) = \frac{s + 0.1}{(s + 0.1)^2 + 16}$$

Into a digital IIR filter by means of the bilinear transformation. The digital filter is to have a resonant frequency of $\omega_r = \pi / 2$.

7. Write short notes on: **(Any two)**

7.2 a) Impulse invariance method for filter design

2.8 b) Frequency response of LTI system.

1 c) Energy and Power Signal

POKHARA UNIVERSITY

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Programme: BE | Full Marks: 100
Course: Digital Signal Analysis and Processing | Pass Marks: 45
Time : 3hrs.

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Attempt all the questions.

1. a) How aliasing is encountered in the processing of discrete-time signals? What can be done to minimize aliasing effect? Explain with the basic elements of digital signal processing where aliasing can be cancelled. 7 1

   b) A digital communication link carries binary-coded words representing samples of an input signal 8 2

   $x_a(t) = 3\cos 5000\pi t + 2\sin 6000\pi t$

   The link is 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?

2. a) Prove that multiplication of a sequence x[n] by e^{jw_0n} is equivalent to a frequency translation of the spectrum X (e^{jw}) by w_0. 7

   b) An LTI, causal, second order discrete-time system is described by the difference equation 8 2

   $$y[n] = 6y[n-1] - 8y[n-1] + 16x[n]$$

   Calculate its step response s[n].

3. a) Prove that necessary and sufficient condition for stability is $\sum_{-\infty}^{\infty} h[k] < \infty$. 7

   b) Why is DFT preferred over DTFT in the analysis of discrete - time signals? Also determine the DFT of the signal x[n] = u[n] - u[n-4] using DIF-FFT algorithm. 8 4

4. a) What is the difference between linear and circular convolution? Find the linear circular convolution of the following sequences: 7 4

   x[n] = {1, 0, 0 ,1} and h[n] = {2,0,2}

Or

Show that multiplication of two discrete time signals results in circular convolution of their DFTs

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

$$H(z) = \frac{1 - 0.8z^{-1} - 0.15z^{-2}}{1 + 0.1z^{-1} - 0.72z^{-2}}$$

Is the system stable?

7 5. a) Design a digital Butterworth low pass filter satisfying the constraints

$$\begin{cases} -3.011dB \le |H(e^{jw})| \le 0dB & 0 \le w \le \frac{\pi}{2} \\ |H(e^{jw})| \le 10.45dB & \frac{3\pi}{4} \le w \le \pi \end{cases}$$

With T = 1sec using bilinear transformation method. Realize the filter using the most convenient realization form.

7 b) Given an analog system

$$H_a(s) = \frac{s^2 + 1.4s + 9.6}{(s + 0.5)(s^2 + 0.4s + 9.4)}$$

Obtain a digital filter by using impulse invariant method. Assume T = 1.

6.4 6. a) Explain briefly the design of linear phase FIR filter by frequency Sampling method with proper example

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

$$0.98 \le |H(e^{j\omega})| \le 1.02, \quad for\ 0 \le |\omega| \le 0.2\pi$$
$$|H(e^{j\omega})| \le 0.03, \quad for\ 0.3\pi \le |\omega| \le \pi$$

7. Write short notes on: (Any two)
   a) Properties of LTI system. 2
   b) Computational complexity of DFT. 4
   c) Limit Cycle Oscillation Effect. ?

POKHARA UNIVERSITY

Level: Bachelor　　Semester: Fall　　Year : 2017
Programme: BE　　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) Define signal processing? Compare between energy signal and power signal. 7 1

   b) State and prove the time-shifting property of DTFS. 2 8

2. a) If the impulse response to a LTI system is given as 7

   $h[n] = \{1\ 2\ 1\}$

   and the output to that system is

   $y[n] = \{1\ 3\ 4\ 3\ 1\}$

   Determine the input signal of that system.

   b) Define causal and non-causal system. Prove that a LTI system is stable if and only if its impulse response is absolutely summable. 8 2

3. a) Define z – transform and Region of Convergence. Find z – transform of following signal $x(n) = a^n \sin w_0 n\, u(n)$. Also determine the ROC of the signal. 7 3

   b) Compute the eight-point DFT of the sequence 8 4

   $$x[n] = \begin{cases} 1, & 0 \le n \le 7 \\ 0, & otherwise \end{cases}$$

   by using the decimation in frequency FFT algorithm.

4. a) Given a three stage lattice filter with coefficients $k_1 = 1/2$, $k_2 = 1/4$ and $k_3 = 1/4$. Determine the FIR filter coefficients for the direct form structure. 7 5

   b) Obtain the direct form realizations for the system described by the following equation: 8 5

   $2y(n) + y(n-1) - 4y(n-5) = x(n) + 3x(n-4)$

5. a) Why ideal lowpass filter cannot be realized in practice? Explain how practical lowpass filter are realized in practice and also explain its 7

effect.

6·3 b) Design a linear FIR filter using Kaiser window to meet the following specifications:

$$0.98 \leq |H(e^{j\omega})| \leq 1.02, \qquad for\ 0 \leq |\omega| \leq 0.19\pi$$
$$|H(e^{j\omega})| \leq 0.02, \qquad for\ 0.21\pi \leq |\omega| \leq \pi$$

7·3 6. a) Use the bilinear transformation to convert the analog filter with system function

$$H(s) = \frac{s + 0.1}{(s + 0.1)^2 + 9}$$

into a digital IIR filter. Select T = 0.1 and compare the location of the zeros in H(z) with the locations of the zeros obtained by applying the impulse invariance method in the conversion of H(s). 1

7·3 b) Design a digital lowpass Butterworth filter using Bilinear Transformation method to meet the following specifications:

Passband attenuation ≤ 1 dB
Stopband attenuation ≥ 30 dB
Passband edge = 1.5 KHz
Stopband edge = 4 KHz
Sampling frequency = 10 KHz

7. Write short notes on: (Any two) 2

1 a) Digital Signal Processing versus Analog Signal Processing

44 b) Time shifting and Frequency shifting properties of Fourier Transform

c·1 c) Gibb's phenomenon

POKHARA UNIVERSITY

Level: Bachelor　　　Semester: Fall　　　Year : 2016
Programme: BE　　　Full Marks: 100
Course: Digital Signal Analysis and Processing　　　Pass Marks: 45
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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) Differentiate between signal analysis and signal processing. What are the advantages of digital signal processing over analog signal processing? Explain. 8 1
   b) Show that the impulse response of a BIBO stable LTI system is absolutely summable. 7 2
2. a) Verify the commutative property of the convolution between two signals x[n] = a^n u[n], 0<a<1 and h[n] = u[n]. 8 2
   b) Find the Z-transform of $x[n] = a^n cos w_0 n\, u[n]$ 7 3
3. a) Find 8-point DFT of the sequence {1,2,3,4,5,4,3,2} using radix-2 decimation in time algorithm. 8 4
   b) Explain the "Multiplication of two DFT's and circular convolution" property of DFT. 7 4
4. a) Draw the lattice ladder diagram for the following system: 8 5
   $H9z) = \frac{1-0.8z^{-1}-0.15z^{-2}}{1+0.2z^{-1}+0.8z^{-2}}$, Is the system stable?
   b) Determine the direct form realisations of the following differential equation: 7 5
   2y[n] = 4y[n-1] - 4y[n-2] + 6y[n-4] + x[n-1] – x[n-3]
5. a) What is Linear phase FIR filter? Explain FIR filter design by Frequency-Sampling method. 8 5.4
   b) Design a linear FIR filter using Kiser window to meet the following specifications: 7 6.3

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

6. a) Design a digital Butterworth lowpass filter using Bilinear 8 7

Transformation method to meet the following specifications:

Passband attenuation≤1.28 dB   Passband edge frequency=200 Hz

Stopband attenuation≥13 db   Stopband edge frequency=350 Hz

Sampling frequency=1 KHz.

6v57 b) Compare between FIR and IIR filter.

7. Write short notes on: **(Any two)**

6'5 a) Remez Exchange Algorithm

3 b) Z-transform and ROC

2 c) Casual and Non-casual system

POKHARA UNIVERSITY

Level: Bachelor | Semester: Spring | Year : 2015
Programme: BE | 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) Write the advantages of digital signals over analog signals. Define Energy and Power signals. 8 1

   b) Show that the impulse response of a BIBO stable LTI system is absolutely summable. 7 2

2. a) Find the convolution between two signals $x[n] = u[n]$ and $h[n] = a^n u[n]$, $0 < a < 1$ 8 2

   b) Find the Z-transform of $x[n] = a^n \cos w_0 n\, u[n]$ 7 3

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

   b) What is Zero Padding ? Find the circular Convolution of $x_1[n] = \{1,2\}$ & $x_2[n] = \{3,2,1\}$ 7 4

4. a) Obtain the parallel form realization of following IIR filter 7 5

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

   b) Convert the following IIR filter into lattice ladder structure 8 5

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

5. a) Design a lowpass filter which will have -3 dB 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 100 samples/sec. 8

   b) Design a linear FIR filter using Kaiser window to meet the following specifications: 7 6/3

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

6. a) Determine H(z) using impulse invariance method for the following analog 7 7.2

filter's transfer function :

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

b) Design a second order discrete-time Butterworth filter with cut-off frequency of 1 KHz and sampling frequency of 10^4 samples/sec by using bilinear transformations. Also plot the poles of the filter.

7. Write short notes on: **(Any two)** 2

a) Region of Convergence

b) Causal and Non-Causal system

c) Gibbs Phenomena