POKHARA UNIVERSITY

Level: Bachelor | Semester: Fall | Year : 2024
Programme: BE | Full Marks : 100
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.

The figures in the margin indicate full marks.

Attempt all the questions.

1. a) Explain the importance of DSP in various fields of engineering and technology. Give a brief account of its applications. 7

   b) State and prove the necessary and sufficient condition for an LTI system to be causal and stable. 8

2. a) 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}

   b) Determine the inverse z-transform of $H(z) = \frac{1+3z^{-1}+2z^{-2}}{1+3.5z^{-1}+1.5z^{-2}}$ for possible ROCs. 8

3. a) Show that z-transform of the sequence $x(n) = -a^n u(-n-1)$ is anti-causal exponential sequence. Also, Find the Z-transform and ROC of: 8

   $x(n) = 2(\frac{5}{6})^n u(-n-1) + 3(\frac{1}{2})^{2n} u(n)$

   Sketch the ROC and pole-zero location.

   b) Define the term zero padding. Find the circulation convolution between $x_1[n] = u[n] + u[n-1] - u[n-3] - u[n-4]$ and $x_2[n] = u[n] - u[n-4]$ using matrix method. 7

4. a) Draw the lattice ladder diagram for the following system: 8

   $H(z) = \frac{1-0.8z^{-1}-0.9z^{-2}}{1+0.2z^{-1}+0.8z^{-2}}$. Is the system stable?

   b) Obtain the Direct form I and Direct form II realization for the systems described by the following equations. 7

   $y(n) = 2x(n) + 0.3x(n-1) + 0.5x(n-2) - 0.7y(n-1) - 0.9y(n-2)$

5. a) Briefly explain the concept of designing analog low pass filter and digital low pass filter with appropriate example. A digital filter has the following impulse response h(n) = {2,4,6,6,4,2}. Is it a linear phase filter? If yes, how? 7

b) Design the symmetric FIR low pass filter for which desired frequency response is expressed as $H_d(\omega) = \begin{cases} e^{-j\omega\tau} \text{ for } |\omega| \le \omega_c \\ 0 \quad elsewhere \end{cases}$ 8

The length of the filter should be 7 and $\omega_c = 1$ radian/sample. Make use of Hanning window function.

6\. a) Determine H(z) using Impulse invariant technique for the analog system function $H_a(s) = \frac{1}{(s+1)(s^2+s+2)}$. 7

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 4kHz. Use the Bilinear transformation technique and obtain H(z) of the desired filter. Take sampling frequency as 10kHz. 8

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

a) Energy Vs Power Signal

b) Frequency response of LTI system

c) Remex exchange algorithm