CSE-417 Digital Signal Processing

Set-01

1. What is DSP? Draw the block diagram of a DSP system.’[13]
2. What do you mean by DSP? List the applications of DSP.’[12,11]
3. Write some advantages of digital over analog signal processing.’[13]
   Or, What are the advantages of DSP over analog signal processing?’[10]
4. Consider a system whose output *y(n)* is related to the input *x(n)* by $y(n) = \sum_{k=-\infty}^{\infty} x(k)\, x(n+k)$
   Determine whether or not the system is---
   (i) linear, *(ii)* shift-invariant, *(iii)* stable, *(iv)* causal.’[13]
5. Check for the following systems are linear, causal, time in variant and static :---
   (i) $y(n) = x(2n)$;
   (ii) $y(n) = \cos(x(n))$;
   (iii) $y(n) = x(n)\cos(x(n))$;
   (iv) $y(n) = x(-n+2)$;
   (v) $y(n) = x(n) + n\,x(n+1)$.'[10]
6. Define the following terms :---
   (i) Discrete time signal;
   (ii) Continuous time signal;
   (iii) Sampling;
   (iv) Quantization.’[13,11]
   Or, What is a continuous and discrete time signal?’[10]
7. *Write the difference between Analog signal and Digital signal*.’[12]
8. Consider the analog signal $x_a(t) = 3\cos 100\pi t$:---
   (i) Determine the minimum sampling rate required to avoid aliasing.
   (ii) Suppose that the signal is sampled at the rate $F_s = 200$ Hz. What is the discrete-time signal obtained after sampling?
   (iii) Suppose that the signal is sampled at the rate $F_s = 75$ Hz. What is the discrete-time signal obtained after sampling?
   (iv) What is the frequency $0<F<F_s/2$ of a sinusoid that yields samples identical to those obtained in part *(iii)*?’[12]
9. How analog to digital conversion of a signal is done?’[12,11]
   Or, Describe the analog-to-digital (A/D) conversion procedure.’[10]
10. Characterize the following systems :---
    (i) $Y(n) = n\,X(n)$;
    (ii) $Y(n) = X(n^2)$.’[11]
11. What is correlation and autocorrelation?’[10]

Set-02

12. What are the properties of convolution? What are the steps involved in calculating convolution sum?’[13,11]
13. Find the convolution of the two finite-length sequences :---
    $x(n) = 0.5\,n[u(n) - u(n-6)]$
    $h(n) = 2\sin\left(\frac{nr}{2}\right)[u(n+3) - u(n-4)]$.’[13]
14. A linear shift-invariant system has a unit sample response $h(n) = u(-n-1)$.
    Find the output if the input is $x(n) = -n3^n u(-n)$.’[13]
15. Prove the distributive property of convolution.’[13]
16. Define the following :---
    (i) Periodic and aperiodic signal; *(ii)* Unit step and ramp signal.’[12]
17. Determine if the following system are linear or non-linear :
    (i) $y(n) = x^2(n)$; (ii) $y(n) = e^{x(n)}$.’[12]
18. Discuss the four steps involving in computing the convolution of *x(k)* and *h(k)*.’[12]
19. What do you mean by LTI system?’[12]
20. What do you mean by LTI system and causality of an LTI system?’[10]
21. Write down the properties of cross co-relation and auto co-relation.’[12]
22. Explain the input-output description of a Discrete Time System.’[11]
23. Determine the output response of a system $y(n)$ to the input signal
    $x(n) = \{0, 3, 2, 1, \underset{\uparrow}{0}, 1, 2, 3, 0\}$ and

$y(n) = \frac{1}{3}[x(n+1) + x(n) + x(n-1)]$.'[11]

24. What are the basic building blocks of discrete time systems?'[10]
25. What is memory system and memoryless system?'[10]

Set-03

26. Write down the properties of ROC.'[13]
27. Define z-transform and ROC of **z-transform**.'[12]
28. What are the properties of ROC in **Z-transform**.'[11]
29. Define : *(i)* **z-transform**, *(ii)* poles, *(iii)* zeroes.'[13]
30. Prove : *(i)* The reversal property of **Z-transform**, *(ii)* Convolution property of **Z-transform**.'[13]
31. Describe the properties of **z-transform** *(i)* Linearity; *(ii)* Time shifting; *(iii)* Convolution.'[12]
32. State and prove the properties of **Z-transform**.'[10]
33. State the convolution property of **Z-transforms**.'[10]
34. Define poles and zeros. Determine the pole-zero of the signal $x(n) = a^n\, a(n)$, $a > 0$.'[12]
35. What are the advantages and disadvantages of FIR filter?'[11]
36. What is the advantage of using normalized frequency in designing filter?'[11]
37. How phase distortion and delay distortion are introduced?'[11]
38. What is the relation between Fourier transform and **Z-transform**?'[10]

Set-04

39. Explain inverse **z-transform**.'[13]
40. Determine the inverse **z-transform** of $x(z) = \dfrac{1}{1-1.5z-1+0.5z-2}$,
    when *(a)* ROC : $|z| > 1$; *(b)* ROC : $|z| < 0.5$.'[12,10]
41. Find the inverse DTFT of $X(e^{jw}) = \dfrac{1}{1-\frac{1}{3}e^{-j10w}}$.'[13]
42. Assume two finite duration sequences $x_1(n)$ and $x_2(n)$ are linearly combined. Let $x_3(n) = a\,x_1(n) + b\,x_2(n)$. What is the DFT of $x_3(n)$?'[13]
43. Why FFT is needed? Calculate the number of multiplications needed in the calculation of DFT using FFT with 32 point sequence.'[13]
44. What is DFT and FFT?'[12]
45. What is DFT and inverse DFT?'[11]
46. Write down FIR filter design procedure.'[12,10]
47. Consider the signal $x(n) = a^n u(n)$, $0 < a < 1$, the spectrum of this signal is sampled at frequencies $W_k = 2\pi k/N$, $k = 0, 1, \ldots, N-1$. Determine the reconstructed spectra for $a = 0.8$ when $N = 5$ and $N = 50$.'[12]
48. Derive the DFT of the sample data sequence $x(n) = \{1, 1, 2, 2, 3, 3\}$ and compute the corresponding amplitude and phase spectrum.'[11]
49. State and prove Parseval's theorem.'[11]
50. What is window function?'[10]
51. Determine the discrete Fourier transform $x(n) = (1, 1, 1, 1)$ and prove $x(n) * h(n) = X(z)\,H(z)$.'[10]
52. Write the various frequency transformation in analog domain.'[10]

Set-05

53. Write down the classification of digital filter.'[13]
54. What is digital filter? How digital filter can be classified?'[11]
    Or, How digital filters can be classified?'[10]
55. Describe the rational of choosing between FIR and IIR filter.'[13]
56. Define digital filter. Describe the advantage and disadvantage of digital filter over analog filter.'[13]
57. Describe filter design steps.'[13]
58. Sketch a simplified block diagram of a digital filter with description.'[12]
59. How can you design a digital filter from analog filter?'[10]
60. How adaptive filter can be used as a noise canceller?'[13,12,11]
61. Write down the characteristics of adaptive filter.'[12]
62. Describe the concept of adaptive filtering. Why it is needed?'[12]
63. Describe Recursive Least Square (RLS) algorithm to design adaptive filter.'[11]
64. State and prove the Complex convolution theorem.'[12]
65. Derive **z-transform** from Laplace transform.'[12]
66. What is **Z-transform**? How is Z transform obtained from Laplace transform?'[11]
67. Mention the advantage of direct and cascade structures.'[11]
68. State the circular time shifting and circular frequency shifting properties of DFT.'[11]

69. What is the relationship between **Z-transform** and DTFT?’[11]
70. What are the different methods of evaluating inverse **Z-transform**?’[11]
71. Sketch the block diagram for the direct form realization and the frequency-sampling realization of the M=32, *a*=0, linear-phase (symmetric) FIR filter which has frequency samples :---

$$H(\frac{2\pi k}{32}) = \begin{cases} 1, k = 0,1,2 \\ \frac{1}{2}, k = 3 \\ 0, k = 4,5,.......,15 \end{cases}$$.’[10]

72. Why impulse invariant method is not preferred in the design of IIR filters other than low pass filter?’[10]
73. What is the necessary and sufficient condition for the linear phase characteristics of a FIR filter?’[10]

Set-06

74. Sketch a general block diagram of an FIR filter. State the general impulse response and transfer function of FIR filter.’[13,11]
75. If *h(n)* is the unit sample response of an ideal low-pass filter with a cutoff frequency $w_c = {}^{r}/_{4}$, find the frequency response of the filter that has a unit sample response *g(n) = h(2n)*.’[13]
76. How phase distortion and delay distortion are introduced?’[13]
77. Describe basic LMS adaptive algorithm with flow chart.’[13,10]
78. Describe Window method for co-efficient calculation of FIR filter.’[12]
79. What do you mean by signal-flow graph?’[12]
80. Describe the Radix-2 algorithm and define Butterfly operation.’[12]
81. How many multiplications and additions are required to compute N-Point DFT using radix 2 FFT?’[11]
82. What is the principle of pole-zero placement method for calculating co-efficients of IIR filter?’[11]
83. Explain the method of design of IIR filters using bilinear transform method.’[10]
84. What do you mean “dead band” of the filter?’[10]