CS2403 - Digital Signal Processing For THIRD IT AND FINAL CSE

POSSIBLE 2 TWO MARKS QUESTIONS(UNIT I-V)

1. What is meant by aliasing? How can it be avoided?
2. Find the energy and power of $x(n) = Ae^{j\omega n} u(n)$.
3. Determine which of the following sequences is periodic, and compute their fundamental period. (a) $Ae^{j7\pi n}$ (b) $sin(3n)$
4. Is the system $y(n) = ln[x(n)]$ is linear and time invariant?
5. Determine Z transform of $x(n) = 5^n u(n)$
6. State sampling theorem.
7. Find the signal energy of $(1/2)^n u(n)$
8. Determine whether the following sinusoids is periodic, if periodic then compute their fundamental period. (a) $cos\ 0.01\pi n$ (b) $sin(\pi 62n/10)$
9. Check whether the system $y(n) = e^{x(n)}$ is linear.
10. Determine Z transform of $x(n) = a^n u(n)$
11. How DFT is differ from DTFT
12. Find DFT of sequence $x(n) = \{1, 1, -2, -2\}$
13. What are the computational saving (both complex multiplication and complex addition) in using N – point FFT algorithm.
14. What do you mean by in – place computation?
15. Differentiate between DIT and DIF FFT algorithm.
16. Write DFT pair of equation
17. List any four properties of DFT.
18. Compute DFT of $x(n) = \{1, -1, 1, -1\}$
19. Calculate % saving in computing through radix – 2, DFT algorithm of DFT coefficients. Assume N = 512.
20. Find the value of W_N^K when N = 8 and K = 2 and also k = 3
21. What are the advantages of FIR filters?
22. What are the desirable characteristics of windows?
23. Define Phase Delay and Group Delay.

24. Draw the Direct form I structure of the FIR filter.
25. Compare FIR and IIR digital filter
26. Draw the ideal gain Vs frequency characteristics of HPF and BPF.
27. What is Gibb's phenomenon?
28. Write the steps involved in FIR filter design.
29. List out the different forms of structural realizations available for realizing a FIR system.
30. Use the backward difference for the derivative and convert the analog filter to digital filter given $H(s)=1/(s^2+16)$
31. State the relationship between the analog and digital frequencies when converting an analog filter using bilinear transformation.
32. Explain the advantage and drawback of Bilinear transformation
33. Explain the term " wrapping effect"
34. Find the transfer function for normalized butterworth filter of order 1 by determining the pole values.
35. Find digital filter equivalent for $H(s)=1/(s+8)$
36. Sketch the mapping of s – plane and z – plane in bilinear transformation.
37. Represent decimal number 0.69 in fixed point representation of length N = 6
38. What is Vocoder.
39. What are the different formats of fixed point representation?
40. State a few applications of adaptive filter

<u>POSSIBLE 16 SIXTEEN MARKS QUESTIONS(UNIT I-V)</u>

41. (i) Find the convolution of the signals $x(n) = (a)^n u(n)$ $(a)^n u(n)$ and $h(n) = (b)^n$ $(b)^n$ $u(n)$. (8)

(ii) Consider a system $y(n) + \frac{11}{44} y(n-1) = x(n) + \frac{11}{22} x(n-1)$. Find transfer function, and impulse response the system. (8)

42. (i) Find inverse Z – transfer of

$$X(Z) = \frac{1}{1-1.5\,Z^{-1}+0.5\,Z^{-2}} \frac{1}{1-1.5\,Z^{-1}+0.5\,Z^{-2}} \quad \text{if}$$

1. ROC : |Z| > 1, (2) ROC : |Z| < 0.5, (3) ROC : 0.5 < |Z| <1 (12).

(ii) Derive expressions to relate Z – transfer and DFT (4)

43.(i)Determine the transfer function, and impulse response of the system $y(n) - \frac{3}{4}\frac{3}{4}\, y(n-1) + \frac{1}{8}\frac{1}{8}\, y(n-2) = x(n) + \frac{1}{3}\frac{1}{3}\, x(n-1)$. (8)

(ii) Find the convolution sum of

$$x(n) = \begin{cases} 1 & n = -2, 0, 1 \\ 2 & n = -1 \\ 0 & otherwise \end{cases}$$

and $h(n) = \delta(n) - \delta(n-1) + \delta(n-2) - \delta(n-3)$. (8)

44. (i) Find the Z transform of (4 + 4)

1. $x(n) = 2^n\, u(n-2)$
2. $x(n) = n^2\, u(n)$

(ii) State and explain the scaling and time delay properties of Z transform. (8)

45. (i) Discuss the properties of DFT. (10)

(ii) State and prove the circular convolution property of DFT. (6)

46. (i) Compute DFT of following sequence (6)

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

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

(ii) Using DFT and IDFT method, perform circular convolution of the sequence $x(n) = \{1, 2, 2, 1\}$ and $h(n) = \{1, 2, 3\}$. (10)

47.Find DFT of the sequence $x(n) = \{ 1, 1, 1, 1, 1, 1, 0, 0\}$ using radix -2 DIF – FFT algorithm. (16)

48. Compute the eight point DFT of the given sequence $x(n) = \{ ½, ½, ½, ½, 0, 0, 0, 0\}$ using radix – 2 DIT - DFT algorithm. (16)

49. (i)Design digital low pass filter using BLT. Given that $H(s) = \frac{1}{(s+1)(s^2+1.732s+1)}$ $H(s) = \frac{1}{(s+1)(s^2+1.732s+1)}$ Assume sampling frequency of 100 rad/ sec. (8)

(ii) Design IIR filter using impulse invariance technique. Given that $H(s) = \frac{1}{s^2+5s+6}H(s) = \frac{1}{s^2+5s+6}$ and implement the resulting digital filter by adder, multipliers and delays. Assume sampling period TT=1sec. (8)

50. (i) Obtain the direct form1, canonic form and parallel form realization structures for the system given by the difference equation

y $y(nn) = -0.1yy(nn-1) + 0.72\, yy(nn-2) + 0.7xx(nn) - 0.252$

$xx(nn-2)$. (10)

(ii) If $H(s) = \frac{1}{(s+1)(s+2)} H(s) = \frac{1}{(s+1)(s+2)}$ find $H(z)H(z)$ using impulse invariant method for sampling frequency of 5 samples/sec (6)

51. Design butterworth filter using bilinear transformation method for the following specifications

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

$|H(e^{j\omega})| \leq 0.2; \qquad 0.6 \leq \omega \leq \pi$ (16)

52. Design an IIR digital low pass butterworth filter to meet the following requirements: Pass band ripple (peak to peak): ≤ 0.5*dB*, Pass band edge: 1.2*kHz*, Stop band attenuation: ≥ 40*dB*, Stop band edge: 2.0 *kHz*, Sampling rate: 8.0 *kHz*. Use bilinear transformation technique. (16)

53. Design a symmetric FIR low pass filter whose desired frequency is given as

H_d $H_d\, \omega\omega$ $(\quad) = \begin{cases} e^{-j\omega\tau} & for\ |\omega| \leq \omega_c \\ 0 & Otherwise \end{cases}$

$\begin{cases} e^{-j\omega\tau} & for\ |\omega| \leq \omega_c \\ 0 & Otherwise \end{cases}$

The length of the filter should be 7 and $\omega_c \omega_c$ = 1 rad / sample. Use rectangular window. (16)

54. (i)For a FIR linear phase digital filter approximating the ideal frequency response

$H_d H_d\ \omega\omega\ (\quad) = \begin{cases} 1, & |\omega| \leq \frac{\pi}{6} \\ 0, & \frac{\pi}{6} \leq |\omega| \leq \pi \end{cases} \begin{cases} 1, & |\omega| \leq \frac{\pi}{6} \\ 0, & \frac{\pi}{6} \leq |\omega| \leq \pi \end{cases}$

Determine the coefficients of a 5 tap filter using rectangular window. (8)

(ii) Determine unit sample response $hh(nn)$ of a linear phase FIR filter of length M = 4 for which the frequency response at $\omega = 0\ \omega = 0$ and $\omega = \pi/2\omega = \pi/2$ is given as $HH_r(0) = 1$ and

$H \quad H_r(\pi/2) = \quad 1/2\pi/2) = \quad 1/2$ (8)

55. (i) Determine the coefficients h(n) of a linear phase FIR filter of length M = 15

which has a symmetric unit sample response and a frequency response (12)

$$Hr\left(\frac{2\pi k}{15}\right)\left(\frac{2\pi k}{15}\right) = \begin{cases} 1 & for \quad k = 0,1,2,3 \\ 0.4 & for \quad k = 4 \\ 0 & for \quad k = 5,6,7 \end{cases} \begin{cases} 1 & for \quad k = 0,1,2,3 \\ 0.4 & for \quad k = 4 \\ 0 & for \quad k = 5,6,7 \end{cases}$$

(ii) State the advantage of floating point representation over fixed points representation.(4)

56. (i)Determine the first 15 coefficients of FIR filters of magnitude specification is given below using frequency sampling method:

$$H \quad H\ e^{j\omega} e^{j\omega} \quad (\qquad) = \begin{cases} 1, & |\omega| < \frac{\pi}{2} \\ 0, & Otherwise \end{cases} \begin{cases} 1, & |\omega| < \frac{\pi}{2} \\ 0, & Otherwise \end{cases}$$

(12)

(ii) Discuss the effect of finite word length on digital filter. (4)

57. With neat diagram and supportive derivation explain multirate signal processing using two techniques. (16)
58. (i)Explain decimation of sampling rate by an integer factor D and derive spectra for decimated signal. (10)

(ii) Discuss on sampling rate conversion of rational factor I/D (6)

59. Write short notes on (a) Image enhancement (b) Speech Processing (c) Musical sound processing and (d) vocoder. (16)
60. What is adaptive filter? With neat block diagram discuss any four applications of adaptive filter. (16)