JYOTHISHMATHI INSTITUTE OF TECHNOLOGY & SCIENCE
Nusthulapur, Karimnagar

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

III - B.Tech II – Semester – ECE

DIGITAL SIGNAL PROCESSING

SUBJECTIVE QUESTION BANK

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1. (a) Define "discrete-time signal sequence".

   (b) Write a note on 'sampling theorem'.
2. Determine whether the DSP system described by the following equations is linear time invariant.
   i. y(n) = a n x(n)
   ii. y(n) = ax(n-1) + ax(n-1) + b x(n-2)
3. State and prove the necessary & sufficient condition for stability of an LTI discrete time system.
4. The system is represented by the difference equation: y(n) = x(n+1) + 3x(n)+ 5x(n-1) Is the system causal, stable, memory less and time invariant? Justify your answer.
5. What are the advantages and disadvantages of digital signal processing over analog signal processing? Explain.
6. Check the given system for: i. Linearity ii. Causality iii. Stability iv. Time invariance.

   System is y(n)=x(n)-x(-n-1)
7. State and prove differentiation in time and convolution in frequency properties of DTFT.

   Test the stability and causality of the given systems y(n)=x(n)-x(-n-1) ii. y(n)= sin[x(n)].
8. State and prove the basic properties of discrete time systems?
9. A discrete time LTI system has impulse response h(n)={1, 3, 2, -1, 1} for -1 _ n _ 3. Using linearity and time invariance property, determine the system output y(n) if the input x(n) is given by x(n) = 2δ(n)-δ(n-1).
10. a) Define an LTI System and show that the output of an LTI system is given by the convolution of Input sequence and impulse response.

    b) Prove that the system defined by the following difference equation is an LTI system

    y(n) = x(n+1)-3x(n)+x(n-1) ; n≥0.
11. a) Write short notes on classification of systems.

    b) Derive BIBO stability criteria to achieve stability of a system.

12. a) Discuss various discrete time sequences.

b) Give the Basic block diagram of Digital Signal Processor.

13. a) Define Linearity, Time Invariant, Stability and Causality.

b) The discrete time system is represented by the following difference equations in which x(n) is input and y(n) is output. $Y(n) = 3y^2(n-1)-nx(n)+4x(n-1)-2x(n-1)$.

14. a) What is Signal Processing and list the advantages, Limitations of Digital Signal Processing. List out some applications of it.

b) Discuss in brief about the classification of signals.

15. (a) Discuss various Discrete time sequences and mention the importance of each.

(b) Define an LTI system and derive the expression for the output response of an LTI system whose input sequence is x(n) and impulse function of the system is h(n).

16. (a) Define the operations of a signal - Time scaling, Amplitude Scaling and folding.

(b) What is an LTI system? Show that an LTI system combined with time scaling property may result in an Time-variant system.

17. (a) Discuss various Discrete time sequences and mention the importance of each.

(b) Define an LTI system and derive the expression for the output response of an LTI system whose input sequence is x(n) and impulse function of the system is h(n).

18. (a) Give the differences between analog and digital system.

(b) Give the block diagram of Analog Signal Processing and compare with Digital signal Processing system and list out the applications of each.

19. (a) Define Frequency response, Magnitude spectrum, phase spectrum and time delay.

(b) Determine the frequency response, magnitude response and phase response of the second order system $y(n)+1/2y(n-1) = x(n)-x(n-1)$.

20. (a) Discuss the effect of ADC Quantization noise on Signal Quality.

(b) Discuss finite word length effects of implementation of FFT algorithm.

21. (a) Define the operations of a signal - Time scaling, Amplitude Scaling and folding.

(b) What is an LTI system? Show that an LTI system combined with time scaling property may result in an Time-variant system.

22. A causal LTI system is defined by the difference equation

    2y(n)-y(n-2)=x(n-1)+3x(n-2)+2x(n-3). Find the frequency response H(e^{jw}), magnitude response and phase response.

    (a) Define the following terms as referred to LTI discrete time system:

    i. Stability

    ii. Causality

    iii. Time invariance

    iv. Linearity.

    (b) Determine whether the following system is

    i. Linear

    ii. Causal

    iii. Stable

    iv. Time invariant

    y (n) = log10 |x (n)| Justify your answer.

23. (a) Determine the impulse response and step response of the causal system given below and discuss on stability y(n)+y(n-1)-2y(n-2)=x(n-1)+2x(n-2)

    (b) Prove that impulse response of an LTI system is absolutely summable for stability of the system.

24. (a) Define "discrete-time signal sequence".

    (b) Write a note on 'sampling theorem'.

    (c) Determine whether the DSP system described by the following equations is linear time invariant.

    i. y(n) = an x(n)

    ii. y(n) = ax(n-1) + ax(n-1) + b x(n-2)

25. (a) A discrete system is given by following difference equation y(n)-5y(n-1) =x(n) + 4x(n-1) where x(n) is the input and y(n) is the out put. Determine its magnitude and phase response as a function of frequency.

    (b) State and prove convolution theorem.

26. (a) Find the convolution of the given two signals graphically: x(n)=u(n)-u(n-5)

h(n)=2[u(n)-u(n-3)]

(b) Verify the result of part (a) by evaluating directly the convolution sum.

27. (a) State and prove the basic properties of discrete time systems?

(b) A discrete time LTI system has impulse response h(n)={1, 3, 2, -1, 1} for −1 <=n <=3. Using linearity and time invariance property, determine the system output y(n) if the input x(n) is given by x(n) = 2δ(n)-δ(n-1).

28. (a) Find the frequency response of an LTI system given its impulse response

i. h(n)= δ(n)+ 6 δ (n-1)+ 3 δ (n-2)

ii. h(n)=$\frac{1}{3^{n+2}}$. u (n − 2)

(b) Find the magnitude, phase and group delay of a system that has a unit sample response h(n)=δ(n)=α.δ(n-1), where α is real.

29. (a) Define a linear-time invariant system in discrete time. Define the terms causality and stability of such systems. Discuss the properties for the following sequences.

y(n) =x(n), for n <= 1

0 for n = 0

x(n + 1), for n <= −1

(b) Find the

i. Impulse response

ii. Output response for a step input applied at n=0 of a discrete time linear time invariant system whose difference equation is given by y (n) = y (n-1) +0.5 y (n-2) +x (n) +x (n-1).

30. (a) Find the impulse and step responses for the given system y(n)+y(n-1) = x(n)-2x(n-1)

(b) Test the following systems for linearity, time invariance, causality and stability.

i. y(n) = $a^{|x(n)|}$

ii. y(n) = sin(2nfπ/F)x(n)

31. (a) Write four advantages of Digital Signal Processing over Analog Signal Processing.

(b) A signal y (n) is governed by the recursive equation y(n) = 2y(n − 1) +δ(n) with y(0) = 4. Find y (-2), y (3). Is the signal bounded or not?

(c) Convolve the two signals x (n) = (1/2)n x(n)and h(n) = u(n)−u(n−10) where u(n) is unit step function.

32. (a) Prove that the convolution in time domain leads to multiplication in frequency domain for discrete time signals

(b) The output y(n) for a linear shit invariant system, with the input x(n) is given by Y(n) = x(n)-2x(n-1)+x(n-2) Compute and sketch the magnitude and phase response of the system |w| <=π

33. (a) Consider a discrete linear time invariant system described by the difference Equation Y(n)-(3/4)y(n-1)+(1/8)y(n-2)=x(n))+(1/3)x(n-1)

Where y(n) is the output and x(n) is the input. Assuming that the system is relaxed initially obtain the unit sample response of the system.

34. (a) Give the advantages and disadvantages of DSP over ASP.

(b) Derive an expression for a Parsevals relation for discrete time periodic signals.

(c) Check whether the following systems are linear.

i. y(n) = 1/N $\sum_{m=0}^{n-1} x(n - m)$

ii. y(n) = [x(n)]2

35. (a) Determine the impulse response for the systems given by the following difference equations.

i. y(n)+3y(n-1)+2y(n-2)=2x(n)-x(n-1)

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

(b) Obtain condition for stability?

36. (a) Consider a LTI system with unit sample response h(n)n = α^{n} u(n) where α is real and 0 < α < 1 . If the input is x(n) = β^{n} u(n), 0 < |β| < 1 ,determine the output f(n) in the form y(n) = (k1 α^{n} n + k2β^{n}u(n) by explicitly evaluating the convolution sum.

(b) Define causality and stability of LTI system and state the conditions for stability.

37. By explicitly evaluating the convolution sum, evaluate the convolution y(n) =x(n)*h(n)of the sequences

h(n) = α^{n} , 0 <= n < N

0, elsewhere

x(n) = β^{n-n0}, n<=no

38. What are the major classifications of the signal?

39. Define discrete time signals and classify them.

40.Define continuous time signals and classify them.

41.Define discrete time unit step &unit impulse.

42.Define continuous time unit step and unit impulse.

43.Define unit ramp signal.

44.Define periodic signal and non-periodic signal.

45.Define even and odd signal?

46.Define Energy and power signal.

47.Define unit pulse function.

48.Define continuous time complex exponential signal.

49.What is continuous time real exponential signal.

50.What is continuous time growing exponential signal?

51.State the BIBO criterion for stability.

52.Find whether the signal given by x (n) = 5cos (6 πn) is periodic .

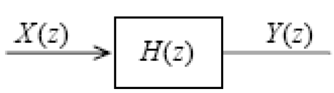
53.For the systems represented by the following functions.

54.Determine whether every system is

(1) stable (2) Causal (3) linear (4) Shift invariant

T[x(n)]= ex(n) (ii) T[x(n)]=ax(n)+6

55.Write down an expression in the z-domain for $Y(z)$ in terms of $X(z)$ and $H(z)$.



56.Write down an expression for $X(z)$ in terms of $x(n)$.

57.Write down an expression for $y(n)$ in terms of $x(n)$ and $h(n)$.

58.Differentiate analog and digital signal processing.

59.Give an example for discrete time signal.

60.Name any four elementary time domain operations for discrete time Domain

61. Write short notes on system and signals?
62. Distinguish between a causal and non causal system,
63. Define even and odd signals.
64. Distinguish between deterministic and random signals.
65. What are multichannel signals?
66. What are the disadvantages of analog signal processing?
67. What are the advantages of digital signal processing?
68. Draw the basic block diagram of DSP system.
69. Differentiate among analog, quantized and digital signal.

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter -1-(pp:2-129)
2. Digital signal processing by P.Ramesh babu Refer Chapter -1-(pp:1.1-1.134)

UNIT –II DISCRETE FOURIER SERIES

1. (a) Prove the following properties

   i. arg[X(K)] = -arg[X((-K)N)RN(K)] ii. Im[X(K)] = -Im[X((-K))NRN(K)]

   (b) If X(K) denotes the N-point DFT of N-Point sequence x(n), show that with N even and if x(n) = x(N-1-n) then X(N/2)=0.

2. (a) Define DFT. properties of DFT.

   (b) Discuss the effects of truncating a sequence x(n) of infinite duration.

   (c) Compute the DFT of x(n) = {-1, 0,-1} with T = 0.5. Plot the DFT sequence suggest a method for improving frequency resolution.

3. (a) Distinguish between DFT and DTFT .

   (b) Consider a sequence x(n) of length L. Consider its DTFT Xd(w) is sampled and N is the number of frequency samples. Discuss the relation between L and N for inverse DTFT = inverse DFT

   comment on the aliasing problem.

   (c) Compute the DFT of x(n) = {1, 0, 0, 0} and compare with Xd(w).

4. (a) If x(n) is a periodic sequence with a period N, also periodic with period 2N.X1(K) denotes the discrete Fourier series coefficient of x(n) with period N and X2(k) denote the discrete Fourier series coefficient of x(n) with period 2N. Determine X2(K) in terms of X1(K).

   (b) Prove the following properties.

   i. Wn Nx(n) ! X ((K + 1))N RN(K) ii. x _ (n) ! X _ ((-K))NRN(K)

5. (a) Compute the discrete Fourier transform of each of the following finite length sequences considered to be of length N.

   i. x(n) = δ(n) ii. x(n) = δ(n - n0) where 0 < n0 < N iii. x(n) = an 0 < n < N - 1

   (b) Let x2(n) be a finite duration sequence of length N and x1(n) = δ(n - n0) where n0 < N. Obtain the circular convolution of two sequences.

6. a) Define DFT and IDFT. State any Four properties of DFT.

   b) Find 8-Point DFT of the given time domain sequence x(n) = {1, 2, 3, 4}.

7. a) Define DFS. State any Four properties of DFS.

b) Find the IDFT of the given sequence x(K) = {2, 2-3j, 2+3j, -2}.

8. a) Define DFS. State any Four properties of DFS.

b) Find the IDFT of the given sequence x(K) = {2, 2-3j, 2+3j, -2}.

9. a) Define Convolution. Compare Linear and Circular Convolution techniques.

b) Find the Linear convolution of the given two sequences x(n)={1,2} and h(n) ={1,2,3} using DFT and IDFT.

10. (a) Compute DFT for the given sequence x(n) = {1, 2, 3, 4}.

(b) Compute linear convolution of two given sequences x(n) ={1, 2, 3} and h(n) = {2, 3}

11. (a) State the properties of DFT of a delayed sequence, Time reversed sequence, circular convolution, Circular frequency shift and circular time shift.

(b) Determine the response of the system whose input x(n) and impulse response h(n) are given by x(n) = {1, 2} and h(n) = {1,2} using DFT and IDFT.

12. (a) Define DFT and IDFT. Prove Circular convolution, Circular correlation and Time reversal properties of DFT.

(b) Find the IDFT of the sequence X(K) = {2, 2-3j, 4, 2+3j}

13. (a) State the properties of DFT of a delayed sequence, Time reversed sequence, circular convolution, Circular frequency shift and circular time shift.

(b) Determine the response of the system whose input x(n) and impulse response h(n) are given by x(n) = {1, 2} and h(n) = {1,2} using DFT and IDFT

14. (a) If x(n) is a periodic sequence with a period N, also periodic with period 2N. X1(K) denotes the discrete Fourier series coefficient of x(n) with period N and X2(k) denote the discrete Fourier series coefficient of x(n) with period 2N. Determine X2(K) in terms of X1(K).

(b) Prove the following properties.

i. Wn Nx(n)□ X ((K + 1))N RN(K)

ii. x * (n) □ X * ((−K))NRN(K)

15. (a) Compute the discrete Fourier transform of each of the following finite length sequences considered to be of length N.

i. $x(n) = \delta(n)$

ii. $x(n) = \delta(n - n0)$ where $0 < n0 < N$

iii. $x(n) = a^n$ $0 <= n <= N - 1$

(b) Let x2(n) be a finite duration sequence of length N and x1(n) = _ (n − n0) where n0 < N. Obtain the circular convolution of two sequences.

16. (a) What is " padding with Zeros ", explain with an example, Explain the effect of padding a sequence of length N with L Zeros (or frequency resolution).

(b) Compute the DFT of the three point sequence $x(n) = \{2, 1, 2\}$. Using the same sequence, compute the 6 point DFT and compare the two DFTs.

17. (a) Define DFT. Guide tow properties of DFT.

(b) Discuss the effects of truncating a sequence x(n) of infinite duration.

(c) Compute the DFT of $X(n) = \{-1, 0, -1\}$ with T = 0.5. Plot the DFT sequence suggest a method for improving frequency resolution. .

18. (a) Define DFT of a sequence x(n). Obtain the relationship between DFT and DTFT.

(b) Consider a sequence $x(n) = \{2, -1, 1, 1\}$ and T = 0.5 compute its DFT and compare it with its DTFT.

19. (a) Compute Discrete Fourier transform of the following finite length sequence considered to be of length N.

i. $x(n) = \delta(n + n0)$ where $0 < n0 < N$

ii. $x(n) = a^n$ where $0 < a < 1$.

(b) If x(n) denotes a finite length sequence of length N, show that

$x((-n))N = x((N - n))N$

20. (a) Define DFT of a sequence. Compute the N - point DFT of the sequence.

$X(n) = Cos(2\pi rn/N)$, $0 <= n <= N - 1$ and $0 <= r <= N - 1$

(b) Explain how DFT can be obtained by sampling DFS for a given sequence.

21. (a) Prove the modulation and time shifting properties of distribute time Fourier transform.

(b) A discrete system is given by following difference equation

$y(n)-5y(n-1) = x(n) + 4x(n-1)$ where x(n) is the input and y(n) is the out put. Determine its magnitude and phase response as a function of frequency

22. (a) State and prove the circular time shifting and frequency shifting properties of the DFT.

(b) Compute the circular convolution of the sequences x1(n) = {1, 2, 0, 1} and

x2(n) = {2, 2, 1, 1} Using DFT approach.

23. (a) Let x(n) and $X(e^{jw})$ represent a sequence and its transform. Determine, in terms of $X(e^{jw})$, the transform of each of the following sequences :

i. g(n) = x(2n)

ii. g(n) = {x(n/2)

24. (a) Distinguish between DFT and DTFT.

(b) Consider a sequence x(n) of length L. Consider its DTFT Xd (w) is sampled and N is the number of frequency samples. Discuss the relation between L and N for inverse DTFT = inverse DFT comment on the aliasing problem.

(c) Compute the DFT of x(n) = { 1, 0, 0, 0 } and compare with Xd (w).

25. Let X(k) be the N-point DFT of the sequence x(n), for 0£ n£ N-1. Define a 2N-point sequence y(n) as x(n/2) n even y(n) = 0 n odd. express the 2N- point DFT of y(n) in terms of X(R).

26. Show that X (e^{jw}) is real and even if, x(n) is real and even.

27. State and prove convolution theorem.

28. Verify parseval's theorem for the DTFT (discrete time Fourier transform)

29. Distinguish between Linear convolution and circular convolution.

30. How will you obtain linear convolution from circular convolution?

31. What is meant by sectioned convolution?

32. What are the two methods used for the sectional convolution?

33. Distinguish between Overlap add and Overlap save method.

34. Distinguish between DFT and DTFT.

35. Distinguish between Fourier series and Fourier transform.

36. Define DTFT.

18. List the properties of DTFT.

39. Define DFT.

40. Define circularly even sequence.

41. Define circularly odd sequence.

42. State circular convolution.

43. State parseval's theorem.

44.State the relation between DTFT and DFT.
45.What is Twiddle factor?
46.Prove the time reversal property of Discrete Fourier Transform.
47.Calculate the number of complex multiplications required for direct computation of DFT and FFT for 64 and 128 point sequences.
48.State and prove the circular frequency shift property of DFT.
49.Relate z-transform and DFT.

50.Compute the DFT of $x(n) = \delta(n - n_o)$.

51.State and prove the Parseval's relation of DFT.

52.What do you mean by the term bit reversal as applied to DFT?

53.Define discrete Fourier series.

54.Draw the basic butterfly diagram of DIF –FFT algorithm.

55.Compute the DFT of $x(n) = a^n$

56.State the time shifting and frequency shifting properties of DFT.

57.Draw the basic butterfly diagram of DIT –FFT algorithm.

58.Determine the 3 point circular convolution of $x(n) = \{1,2,3\}$ and $h(n) = \{0.5,0,1\}$

59.What is FFT and what are its advantages?

60. Distinguish between DFT and DTFT (Fourier transform)

61.What is the basic operation of DIT –FFT algorithm?

62.What is zero padding? What are its uses?

63.State and prove Parseval's relation for DFT.

64.Draw the flow graph of radix – 2 DIF - FFT algorithm for N= 4

65.What do you mean by bit reversal in DFT?

66.Write the periodicity and symmetry property of twiddle factor.

67.Give the relationship between z-domain and frequency domain.

68.Distinguish between discrete Fourier series and discrete Fourier transform.

69.What is the relationship between Fourier series co-efficient of a periodic sequence and DFT?

70.Establish the relation between DFT and z-transform.

71.Define DFT pair.

72.Define Gibbs phenomenon.

73.How many multiplications and additions are required to compute N-point DFT using radix – 2 FFT?

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter -7-(pp:449-488)
2. Digital signal processing by P.Ramesh babu Refer Chapter -3-(pp:3.1-3.53)

UNIT –III :FAST FOURIER TRANSFORMS

1. Why FFT is needed?
2. What is the main advantage of FFT?
3. What is FFT?
4, what is meant by Radix-2 FFT?
5. What is decimation-in-time algorithm?
6. What is decimation in frequency algorithm?
7. What are the differences and similarities between DIF and DIT algorithm?
8. What is the basic operation of DIT algorithm?
9. What is the basic operation of DIF algorithm?
10. What are the applications of FFT algorithms?
11. Draw the flow graph of a two point DFT for a decimation-in-time decomposition.
12. Draw the flow graph of a two point radix-2 DIF FFT.

13.An 8 point sequence is given by x(n) = {2,2,2,2,1,1,1,1}. Compute 8 point DFT of x(n) by
(a) radix - 2 DIT FFT (b) radix - 2 DIF FFT Also sketch magnitude and phase spectrum.

14. (a) Implement the decimation in time FFT algorithm for N=16.
(b) In the above Question how many non - trivial multiplications are required?

15. (a) Let x(n) be a real valued sequence with N-points and Let X(K) represent its DFT , with real and imaginary parts denoted by XR(K) and XI(K) respectively. So that X(K) = XR(K) + JXI(K). Now show that if x(n) is real, XR(K) is even and XI(K) is odd.
(b) Compute the FFT of the sequence x(n) = { 1, 0, 0, 0, 0, 0, 0, 0 }

16. a) Implement the Decimation in frequency FFT algorithm of N-point DFT where N-8. Also explain the steps involved in this algorithm.
(b) Compute the FFT for the sequence x(n) = { 1, 1, 1, 1, 1, 1, 1, 1 }

17. (a) Give the steps involved in implementing Radix -2, DIT FFT algorithm.
(b) Compute the 8-point DFT of x(n) = Cos(PI/2)n 0 < n< 7 using Radix - 2 , DIT algorithm.

18. a) Derive the expressions for computing the FFT using DIT algorithm and hence draw the standard butterfly structure.

b) Compare the computational complexity of FFT and DFT.

19. a) Find X(K) of the given sequence x(n) = { 1,2,3,4,4,3,2,1}using DIT- FFT algorithm.

b) Compare the computational complexity of FFT and DFT.

20.a) Find IFFT of the given X(K) = { 1,2,3,4,4,3,2,1}using DIF algorithm

b) Bring out the relationship between DFT and Z-transform.

21.a) Develop DIT-FFT algorithm and draw signal flow graphs for decomposing the DFT for N=6 by considering the factors for N = 6 = 2.3.

b) Bring out the relationship between DFT and Z-transform.

22. (a) Determine the relationship between DFT and Fourier transform of an aperiodic sequence.

(b) Perform Linear convolution of the two sequences x(n) = {12, 3, -1, -2, -3, 4, 5, 6} and h(n) = {2, 1, -1} using over-lap save method and verify the result using Over-lap add method.

23.(a) What is DIF algorithm? Give the mathematical analysis of DIF algorithm using Radix-2 In-place algorithm.

(b) Compute 8-point DFT of the given sequence x(n) = {1, 2, 1, 2, 1, 2, 1, 2} using DIF FFT algorithm.

24. (a) Compare the computational complexity of DFT and FFT.

(b) An 8 point sequence is given by x(n) = {1, 2, 1, 2, 1, 2, 1, 2}. Compute 8 point DFT of x(n) using Radix-2 DIT FFT.

25. (a) Give the Basic buttery structures of computing DFT using DIT and DIF algorithm.

(b) Compute the IFFT for the sequence X(K) ={0, 1, 2, 3, 0, 0, 0, 0}, using DIT algorithm.

26. (a) Compare DIT and DIF FFT algorithms.

(b) Develop the signal flow graph in computing 16-point FFT using DIT-FFT algorithm.

27. (a) Give the Basic buttery structures of computing DFT using DIT and DIF algorithm.

(b) Compute the IFFT for the sequence X(K) ={0, 1, 2, 3, 0, 0, 0, 0}, using DIT

28. (a) Draw the butterfly line diagram for 8 - point FFT calculation and briefly explain. Use decimation -in-time algorithm.

(b) What is FFT? Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32 point sequence.

29. (a) Explain the inverse FFT algorithm to compute inverse DFT of a N=8. Draw the flow

graph for the same.

(b) Compute the FFT for the sequence { 1, 0, 0, 0, 0, 0, 0, 0 }

30. An 8 point sequence is given by x(n) = {2,2,2,2,1,1,1,1}. Compute 8 point DFT of x(n) by

(a) radix - 2 D I T F F T

(b) radix - 2 D I F FF T Also sketch magnitude and phase spectrum.

31. (a) Implement the decimation in time FFT algorithm for N=16.

(b) In the above Question how many non - trivial multiplications are required.

32. (a) Implement the Decimation in frequency FFT algorithm of N-point DFT where N=8. Also explain the steps involved in this algorithm.

(b) Compute the FFT for the sequence x(n) = { 1, 1, 1, 1, 1, 1, 1, 1 }

33. (a) Draw the butterfly line diagram for 8 - point FFT calculation and briefly explain. Use decimation -in-time algorithm.

(b) What is FFT? Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32 point sequence.

34. What is the computational efficiency of FFT radix-2 DIT algorithm for an N-point sequence? Compare it with that of direct DFT computation.

35. Give the steps involved in implementing radix-2 DIT-FFT algorithm.

36. A designer has available a number of 8-point FFT chips. Show explicitly how he should connect three such chips in order to compute a 24-point DFT

37. Evaluate the 8-point DFT for the following sequence using DIT-FFT algorithm.

X1(n)= 1 for -3<=n<=3 0 otherwise

38. Develop a radix-2 DIF-FFT algorithm for evaluating the DFT for N=16 and hence determine the 16-point DFT of the sequence

x(n)={0,1,0,1,0,1,0,1,0,1,0,1,0,1,0, 1}

40. Compute the 16-point DFT of x(n)=cos_/2 n 0<=n<=15 using radix-2 DIF algorithm.

41. Develop a Radix – 2, DIF/FFT algorithm for an 8-Point sequence.

42. (a) Compare Decimation in time and decimation- in frequency FFT computation.

(b) Compute the FFT of x(n) = (1,2,3,4,4,3,2,1) using DIT FFT algorithm.

43. Find 4-point DFT of the following sequences

(a) x(n)={1,-1,0,0}

(b) x(n)={1,1,-2,-2} (AU DEC 06)

(c) x(n)=2n

(d) x(n)=sin(nΠ/2)

44. Find 8-point DFT of the following sequences

(a) x(n)={1,1,1,1,0,0,0,0}

(b) x(n)={1,2,1,2}

45. Determine IDFT of the following

(a)X(k)={1,1-j2,-1,1+j2}

(b)X(k)={1,0,1,0}

(c)X(k)={1,-2-j,0,-2+j}

46 .In a LTI system, the input X(n) ={1, 2, 1, 2} and the impulse response h(n) = {-1, 1, -1, 1}. Determine the impulse response of the LTI system by Radix – 2 DIF FFT algorithm.

47. The unit impulse response of an FIR digital filter is {1, 2, 1}. Illustrate the Overlap save and Add method to determine its response to the input sequence {1, 0, 2, 0, -1, 1, 0, 0, 1, 2, -1}

48. Determine the FFT of the following sequence using DIT and DIF Radix-2 algorithm. { 1, -1, 1, -1, 1, -1, 1, -1} Also draw the butterfly structure and mention the intermediate values.

49. i) Perform Linear Convolution of the following sequences by Overlap and Save method. X(n) = { -1, 1, 2, -1, 1, 2, -1, 1, -1} and h(n) = {2, 1, -2}

ii) Find the N-Point DFT of {1, 2, 3, 4}

50. i) Perform circular convolution of the sequence using DFT and IDFT technique. $x_1(n)$ = {2, 1,2,1} x_2 (n) = {0,1,2,3}

ii) Compute the DFT of the sequence x(n) = {1,1,1,1,1,1,0,0}

51. From the first principles obtain the signal flow graph for computing 8 – point DFT using radix-2 DIT FFT algorithm. Using the above compute the DFT of sequence x(n) = {0.5,0.5,0.5,0.5,0,0,0,0}

52. Determine the 8-point DFT of the sequence x(n) = {0,0,1,1,1,0,0,0}

53. Compute the output using 8 point DIT – FFT algorithm for the sequence
$x(n) = \{1,2,3,4,5,6,7,8\}$

54. Derive DIF – FFT algorithm. Draw its basic butterfly structure and compute the DFT

$x(n) = (-1)^n$ using radix 2 DIF – FFT algorithm.

55. From the first principles obtain the signal flow graph for computing 8 – point DFT using radix-2 DIF-FFT algorithm. An 8 point sequence is given by $x(n)=\{2,2,2,2,1,1,1,1\}$ compute its 8 point DFT of x(n) by radix-2 DIF-FFT

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter -8-(pp:511-536)
2. Digital signal processing by P.Ramesh babu Refer Chapter -4-(pp:4.1-4.29)

UNIT–IV:REALIZATION OF DIGITAL FILTERS

1. (a) With reference to Z-transform, state and prove the initial and final value theorem.

   (b) Determine the causal signal x(n) having the Z-transform $X(Z) = Z^2+Z/(Z-1/2)^2 (Z-1/4)$

2. (a) Explain how the analysis of discrete time invariant system can be obtained using convolution properties of Z transform.

   (b) Determine the impulse response of the system described by the difference equation $y(n)-3y(n-1)-4y(n-2)=x(n)+2x(n-1)$ using Z transform.

3. (a) An LSI system is characterized by: $y(n) = 5/6y(n-1)-1/6y(n-2)+x(n)$. Find the system function and determine its response to the input signal $x(n) =\delta(n) - 1/3\delta(n - 1)$.

   (b) Write about Jury's criterion.

4. Find the transfer function of a system whose input is: $x(n) = (1/2)^n u(n)-1/4 (1/2)^n-1u(n-1)$ and The output is $y(n) = (1/3)^n u(n)$. Is the system stable? Realize the system with minimum possible amount of memory. Determine its impulse response.

5. (a) An LTI system is described by the equation $y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2)$. Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane.

   (b) Define stable and unstable system. Test the condition for stability of the first-order IIR filter governed by the equation $y(n)=x(n)+bx(n-1)$.

6. Discuss and draw various IIR realization structures like Direct form – I, Direct form-II, Parallel and cascade forms for the difference equation given by $y(n) = - 3/8\ Y(n-1) + 3/32\ y(n-2) + 1/64\ y(n-3) + x(n) + 3\ x(n-1) + 2\ x(n-2)$.

7 .What are the various basic building blocks in realization of Digital Systems and hence discuss transposed form realization structures.

8. a) Define Z-Transform and List out its properties.

   b) Discuss Direct form, Cascade and Linear phase realization structures of FIR filters.

9. a) Discuss transposed form structures with an example.

   b) Discuss Direct form, Cascade realization structures of FIR filters.

10. (a) Discuss the effects due to finite word length in Direct form - I and II structures.

(b) Discuss the effect of quantization of coefficients in FIR filters.

11. Obtain all the possible realization structures of the following transfer function.

$H(Z) = 1 – b\cos\Theta Z^{-1}/(1 + aZ^{-1})(1- 2b\cos\Theta Z^{-1}+b^2z^{-2})$:

12. (a) Determine the impulse response h(n) for the system described by the second order difference equation y(n)-4y(n-1)+4y(n-2) = x(n-1).

(b) An LTI system is described by the equation y(n) = x(n)+0.8x(n-1)+0.7x(n-2)- 0.45y(n-2). Determine the transfer function of the system. Sketch its poles and zeros on the Z-plane.

13. (a) Define Z- Transform. State any four properties of Z- transform.

(b) A system has an impulse response h(n) = {1,2,3} and output response y(n)= {1, 1, 2,-1, 3}. Determine the input sequence.

14. (a) Determine the impulse response h(n) for he system described by the second order difference equation y(n)-4y(n-1)+4y(n-2) = x(n-1).

(b) An LTI system is described by the equation y(n) = x(n)+0.8x(n-1)+0.7x(n-2)- 0.45y(n-2). Determine the transfer function of the system. Sketch its poles and zeros on the Z-plane.

15. (a) Define Z- Transform. State any four properties of Z- transform.

(b) A system has an impulse response h(n) = {1,2,3} and output response y(n)= {1, 1, 2,-1, 3}. Determine the input sequence.

16. An LTI system is described by the equation y(n)=x(n)+0.81x(n-1)-0.81x(n-2)- 0.45y(n-2) Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane.

17. (a) With reference to Z-transform, state the initial and final value theorem.

(b) Determine the causal signal x(n) having the Z-transform.

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

18. (a) Explain how the analysis of discrete time invariant system can be obtained using convolution properties of Z transform.

(b) Determine the impulse response of the system described by the difference equation y(n)- 3y(n-1)-4y(n-2)=x(n)+2x(n-1) using Z transform.

19. Define stable and unstable system. Test the condition for stability of the first-order IIR filter governed by the equation $y(n)=x(n)+bx(n-1)$.

20. (a) Determine the frequency response , magnitude response and phase response for the system given by $y(n) - 3/4y(n - 1) +1/8y(n - 2) = x(n) - x(n - 1)$

(b) A causal LTI system is described by the difference equation $y(n) = y(n-1)+y(n-2)+x(n-1)$, where $x(n)$ is the input and $y(n)$ is the output. Find

i. The system function $H(Z)=Y(Z)/X(Z)$ for the system, plot the poles and zeroes of $H(Z)$ and indicate the region of convergence.

ii. The unit sample response of the system.

iii. Is this system stable or not?

21. (a) Obtain the cascade and parallel form realization of the LTI system governed by the equation.

(b) Compare cascade and performance of direct and canonic forms.

22. (a) Explain the factors that influence the choice of structure for realization of a LTI system.

(b) An LTI system is described by the difference equation $y(n) = a1y(n - 1)+ x(n) + b1x(n - 1)$ Realize it in direct form I structure and convert it to direct form II structure.

23. (a) A causal system is represented by the following difference equation.

$y(n) + 1/4y(n - 1) = x(n) + 1/2x(n - 1)$ Find the system transfer function $H(Z)$, unit sample response and frequency response of the system

(b) Realize $H(Z) = 1+ 1/2Z^{-1}+ 1/8Z^{-2}+ 3/4Z^{-3}+ 1/8Z^{-4}+ 1/2Z^{-5}+Z^{-6}$ with minimum number of multipliers.

24. (a) Explain how the analysis of discrete time invariant system can be obtained using convolution properties of Z transform.

(b) Determine the impulse response of the system described by the difference equation $y(n)- 3y(n-1)-4y(n-2)=x(n)+2x(n-1)$ using Z transform.

25. a) A second order discrete time system is characterized by the difference equation $y(n) - 0.1y(n-1) - 0.02y(n - 2) = 2x(n) - x(n-1)$ .Determine $y(n)$ for $n^3 0$ when $x(n) = u(n)$ and the initial conditions are $y(-1) = -10$ and $y(-2) = 5$.

b) State the conditions for a digital filter to be causal and stable.

26. (a) Explain the different structures for realization of IIR system. and explain how conversion can be made from direct form I structure to direct form II structure.

(b) Realize the given system in cascade and parallel form

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

27. Realize the following IIR system by cascade and parallel forms.

$y(n) + 1/4y(n-1) - 1/8y(n-2) = x(n) - 2x(n-1) + x(n-2)$

28. (a) Explain the parallel form realization for IIR system and obtain the direct form I, direct form II realization of the LTI systems governed by the equation.

$y(n) = -3/8y(n-1) + 3/32y(n-2) + 1/64y(n-3) + x(n) + 3x(n-1) + 2x(n-2)$

(b) Compare cascade and parallel form relations.

29. List the different types of structures for realizing FIR system and determine the direct form-I, direct form II of the following LTI system

$y(n)=-0.5y(n-1)+0.25y(n-2)+0.125y(n-3)+x(n)+0.5x(n-1)+0.75x(n-2)$

30. Explain the structures for realization of FIR system and draw the direct form structure of the FIR system described by the transfer function

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

31. Realize the FIR transfer function

$H(z) = (1+0.8Z^{-1})5 = 1+4Z^{-1} +6.4Z^{-2} +5.12Z^{-3} +2.048Z^{-4} +0.32768Z^{-5}$ in the following forms.

(a) Two different direct forms.

(b) Cascade of first-order sections

(c) Cascade of one first - order and two second order sections and

(d) Cascade of one second - order and one third order sections.

Compare the computational complexity of each and above realizations.

32. (a) Write the difference equations for FIR and IIR system and hence derive the transfer function of FIR and IIR system.

(b) Realize the following system with minimum number of multipliers

$H(Z) = 0.5+0.75Z^{-1}+0.8Z^{-2}+0.9Z^{-3}+2Z^{-4}+0.9Z^{-5}+0.8Z^{-6}+0.75Z^{-7}+0.5Z^{-8}$

33. With reference to Z-transform, state and prove the initial and final value theorem.

34. Determine the causal signal x(n) having the Z-transform $X(Z) = Z^2+Z/(Z-1/2)^2\ (Z-1/4)$

35. Explain how the analysis of discrete time invariant system can be obtained using convolution properties of Z transform.

36. Determine the impulse response of the system described by the difference equation y(n)-3y(n-1)-

4y(n-2)=x(n)+2x(n-1) using Z transform.

37. An LTI system is characterized by: y(n) = 5/6y(n-1)- 1/6y(n-2)+x(n). Find the system function and determine its response to the input signal x(n) =δ(n) – 1/3δ(n - 1).

38. Write about Jury's criterion.

39. Find the transfer function of a system whose input is: $x(n) = (1/2)^n u(n) - 1/4 (1/2)^{n-1} u(n-1)$ and the out put is $y(n) = (1/3)^n u(n)$. Is the system stable? Realize the system with minimum possible amount of memory. Determine its impulse response.

40. An LTI system is described by the equation y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2). Determine the transfer function of the system. Sketch the polesand zeroes on the Z-plane.

41. Define stable and unstable system. Test the condition for stability of the first-order IIR filter governed by the equation y(n)=x(n)+bx(n-1).

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter '9' page no's(563-594)
2. Digital signal processing by P.Ramesh babu Refer Chapter '5' page no's(5.54-5.76)

UNIT-V:IIR DIGITAL FILTERS

1. Mention the need for impulse invariant method.
2. Draw the direct form structure for the following difference equation. Y(n) = Y(n-1) + 5Y(n-2) + X(n) + 2X(n-1)
3. What is prewarping?
4. Write the advantages of bilinear transformation method.
5. Compare the impulse invariance and bilinear transformation methods.
6. Sketch the magnitude response of Type - I and II Chebyshev filter.
7. An analog filter has a transfer function H(s) = 1 / s+2. Using impulse invariance method, obtain pole location for the corresponding digital filter with T = 0.1s.
8. What is frequency warping in bilinear transformation?
9. Find the digital transfer function H (z) by using impulse invariance method for the analog transfer function H(s) = 1 / s+2.
10. What are the different structures of realization of IIR filters?
11. What are the methods used to transform analog to digital filters?
12. Draw the direct form – I structure of IIR filter.
13. Draw the direct form – II structure of IIR filter.
14. Draw the cascade form realization structure of IIR filter.
15. Draw the parallel form realization structure of IIR filter.
16. When cascade form realization structure is preferred in filters?
17. Give the magnitude function Butterworth filter. What is the effect of varying the order of N on magnitude and phase response?
18. List out the properties of Butterworth low pass filter.
19. List out the properties of Chebyshev filter.
20. Give the Chebyshev filters transfer function and draw its magnitude response.

21. Give the equation for the order 'N' and cut off frequency Ω_c of Butterworth filter.
22. Why impulse invariance method is not preferred in the design of IIR filter other than low pass filters?
23. What are the advantages and disadvantages IIR filters?
24. Compare Butterworth and Chebyshev filters.
25. What are the parameters that can be obtained from the Chebyshev filter specification?
26. Give the bilinear transform equation between s-plane and z-plane.
27. What are the properties of bilinear transformation?
28. What is the main disadvantage of direct-form realization?
29. Realize $y(n) + y(n+1) + \frac{1}{4} y(n-2) = x(n)$ in cascade form network.
30. What is the advantage of cascade realization?
31. i) Explain in detail the design procedures of IIR digital filters using Impulse Invariant method.

    ii) State the need for realization of filters in different forms.

32. Design a digital Butterworth filter satisfying the following constraints with T = 1 sec using Bilinear Transformation.

    $0.707 \leq |\mathbf{H(e^{j\omega})}| \leq 1$ for $0 \leq \omega \leq \pi / 2$

    $|\mathbf{H(e^{j\omega})}| \leq 0.2$ for $3\pi / 4 \leq \omega \leq \pi$

33. Design a Chebyshev IIR Digital filter to meet the following desired frequency response $H(\omega)$ by using Impulse Invariant techniques with sampling period T = 1 sec.

    $0.9 \leq |\mathbf{H(\omega)}| \leq 1$; $0 \leq \omega \leq \pi / 4$

    $0 \leq |\mathbf{H(\omega)}| \leq 0.25$; $\pi / 2 \leq \omega \leq \pi$

34. i) Obtain the Cascade and Parallel realization of the system described by

    $Y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$

    ii) Write short notes on frequency transformation in analog domain.

35. An IIR digital low-pass filter is required to meet the following specifications:

Pass band ripple (or peak to peak ripple) $\leq$ 0.5 dB

Pass band edge: 1.2 KHz

Stop band attenuation: ≥ 40dB

Stop band edge : 2.0 KHz

Sample rate: 8.0 KHz

Determine the required filter order for

i) Digital Butterworth filter

ii) Digital Chebyshev filter

36. Convert the following analog transfer function $H(s) = (s+0.2) / [(s+0.2)^2 + 4]$ into equivalent digital transfer function H (z) by using impulse invariance method assuming T= 1 sec.
37. Convert the following analog transfer function H(s) = 1 / (s+2) (s+4) into equivalent digital transfer function H (z) by using bilinear transformation with T = 0.5 sec.

38. Convert the following analog transfer function H(s) = 2/ (s+1) (s+3) into equivalent digital transfer function H (z) by using bilinear transformation with T = 0.1sec. Draw the direct form – II realization of digital filter.
39. Explain impulse invariance method of digital filter design.
40. Derive an expression between s- domain and z- domain using bilinear transformation. Explain frequency warping.
41. Justify the statement IIR filter is less stable and give reasons for it.

42. Find filter order for following specifications p0.5 _ |H (ejw)| =0.1:10 _< ω _< π/2 |

H (ejw)| = 0.2

3π/4 <_ ω _< π With T = 1 sec. use Impulse Invariant method.

43. Design Butterworth filter with at least 66dB attenuation at = 2000π rad /sec and 3dB attenuation at1000π rad /sec . Also realize the filter with suitable technique. Use Impulse Invariant method for analog to digital conversion.
44. With Impulse Invarience method a first order pole in Ha (s) at s=sk is mapped to a pole in H(Z)at Z = eskTs as1/(s - sk) ==> 1/(1 - eskTsZ-1)Determine how a second order will be mapped.
45. Discuss the concept of frequency transformation in analog domain .

46. Determine the system function H(Z) of the lowest order chebyshev digital filter that meets following specifications.

(a) 1dB ripple in the pass band 0 _ |ω| _ 0.3π

(b) At least 60 dB attenuation in the stop band

0.35π _ |ω| _ π . Use bilinear transformation method.

47. What is frequency warping ? How it will arise?

48. Compare Impulse invariant and bilinear transformation methods.

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter '10.3' page no's(701-727)
2. Digital signal processing by P.Ramesh babu Refer Chapter '5' page no's(4.43-5.54)

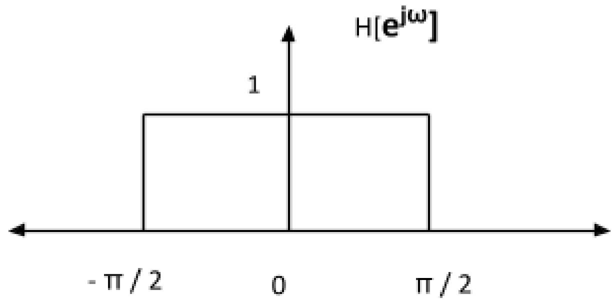
UNIT-VI:FIR DIGITAL FILTERS

1. What are linear phase characteristics in FIR filters?
2. Draw the shapes of any four window functions.
3. Why do we use window functions in FIR filter design?
4. Draw the block diagram of poly phase structure.
5. What is the effect of having abrupt discontinuity in frequency response of FIR filters?
6. What are the characteristic features of FIR filters?
7. What are the different structures of realization of FIR filters?
8. Write the equation of Hamming window.
9. Write the equation of Hanning window.
10. Write the equation of Blackman window.
11. Draw the basic FIR filter structure.
12. Distinguish between FIR and IIR filters.
13. Compare analog and digital filters.
14. Why FIR filters are always stable?
15. State the condition for a digital filter to be causal and stable.
16. What are the desirable characteristics of windows?
17. What are the advantages and disadvantages FIR filters?
18. What are the desirable and undesirable features of FIR filters?
19. What are the design techniques of designing FIR filters?
20. Draw the direct form realization of FIR system.
21. What do you understand by linear phase response?
22. For what kind of applications, the anti-symmetrical impulse response can be used?
23. For what kind of applications, the symmetrical impulse response can be used?
24. Draw the direct form realization of a linear phase FIR system for N even.
25. Draw the direct form realization of a linear phase FIR system for N odd.
26. What is Gibbs phenomenon?
27. What are the desirable characteristics of the window?
28. What is the principle of designing FIR filter using frequency sampling method?
29. For what type of filters frequency sampling method is suitable?
30. Give the equation of the frequency sampling realization of FIR filter.
31. State the condition for linear phase in FIR filters for symmetric and anti -symmetric response.

32. i) Explain Symmetric and Anti-symmetric methods of FIR filter design.
ii) Explain the design of Linear phase FIR filters by the frequency sampling method.

33. Explain the design procedures of FIR digital filters using Hamming & Hanning window.

34. Design a LPF for the following purpose. Use Hamming Window N=7. Fig.1



35. i) What are linear phase characteristics? (4)

ii) Explain the design procedures of FIR filter using Blackman window. (12)

36. With suitable examples, describe the realization of linear phase FIR filters.

37. Design a low pass filter using Hamming window for N=7 for the desired frequency Response

$D(\omega) = e^{j3\omega}$ for $-3\pi / 4 \leq \omega \leq 3\pi / 4$

$= 0$ for $3\pi / 4 \leq \omega \leq \pi$

38. Design a LPF with

$H_d(\omega) = e^{-j3\omega}$ for $-3\pi/4 \leq \omega \leq 3\pi/4$

$= 0$ for $3\pi/4 \leq \omega \leq \pi$

Using a Hamming window with N= 7

39. Using frequency sampling method, design a bandpass filter with the following specifications.

Sampling frequency, F = 8000 Hz

Cut off frequencies, f_{c1}= 1000 Hz and f_{c2} = 3000 Hz.

Determine the filter coefficients for N = 7.

40. Design an ideal low pass filter with a frequency response

$$H_d(e^{j\omega}) = 1 \quad \text{for} \quad -\pi/2 \leq |\omega| \leq \pi/2$$

$$= 0 \quad \text{for} \quad \pi/2 \leq |\omega| \leq \pi$$

Find the values of h(n) for N=11 and H(z)

41. Consider an FIR filter with system function

$$H(z) = 1 + 2.88\, z^{-1} + 3.4048\, z^{-2} + 1.74\, z^{-3} + .4z^{-4}$$

sketch the direct form I , II and determine in detail the corresponding input – output equations.

Is the system minimum phase?

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter '10.2' page no's(660- 700)
2. Digital signal processing by P.Ramesh babu Refer Chapter '6' page no's(6.1-6.95)

UNIT VII:MULTIRATE DIGITAL SIGNAL PROCESSING:

1. List the applications of multirate signal processing.
2. State the Nyquist sampling theorem.
3. Distinguish Upsampling & Downsampling.
4. What is aliasing? Mention the cause producing aliasing effect.
5. Define decimation and interpolation.
6. What is the need for multirate signal processing?
7. What is multirate signal processing?
8. Define downsampling.
9. What is meant by upsampling?
10. If the spectrum of a sequence x(n) is $X(e^{j\omega})$, the what is the spectrum of a signal down sampled by a factor of 2?
11. If the z-transform of a sequence x(n) is X(z) then what is the Z-tranform of a sequence down sampled by a factor M?
12. If the z-transform of a sequence x(n) is X(z) then what is the z-tranform of a sequence up sampled by a factor L?
13. What is the need for antialiasing filter prior to downsampling?
14. What is the need for anti-aliasing filter after up-sampling a signal?
15. What are the characteristics of a narrow band low pass filter?
16. What are types of filter banks?
17. Draw the structure of an analysis filter bank.
18. Draw the structure of a synthesis filter bank.
19. Draw the structure of a two-channel quadrature mirror filter bank.

20. What are the sections of a two channel QMF bank?

21. i) Write short notes on Sampling & Sampling Rate alteration.
ii) Explain the different interpolation techniques with suitable examples.

22. i) Explain the principle of sampling rate conversion for multirate signal processing.

ii) What is Nyquist sampling theorem? Explain Decimation by a factor of 'D' and Interpolation by a factor of 'L' with suitable examples.

23. Explain in detail the poly phase structures for decimation and interpolation filters.

24. Explain the design of narrow band filters in detail.

25. With necessary equations and diagrams, discuss about the interpolation and decimation in multirate signal processing.

26. Design one-stage and two-stage interpolators to meet the following specification:
I = 20
Input sampling rate : 10000 Hz
Pass band : $0 \leq F \leq 90$
Transition band : $90 \leq F \leq 100$
Ripple : $\boldsymbol{\delta_1 = 10^{-2}}$ and $\boldsymbol{\delta_2 = 10^{-3}}$

27. i) Explain the design procedures of narrow band filters.
ii) What is aliasing? Explain the aliasing effect in digital filters with suitable waveforms. Mention the methods of eliminating the aliasing effect.

List the applications of multirate signal processing.

28. Distinguish Upsampling & Downsampling.

29. What is aliasing? Mention the cause producing aliasing effect.

30. Define decimation and interpolation.

31. What is the need for multirate signal processing?

32. What is multirate signal processing?

33. Define down sampling.

34. What is meant by upsampling?

35.If the spectrum of a sequence x(n) is $X(e^{j\omega})$, the what is the spectrum of a signal down sampled by a factor of 2?

36.If the z-transform of a sequence x(n) is X(z) then what is the z-tranform of a sequence down sampled by a factor M?

37. If the z-transform of a sequence x(n) is X(z) then what is the z-tranform of a sequence upsampled by a factor L?

38. What is the need for anti aliasing filter prior to down sampling?

39.What is the need for anti-aliasing filter after upsamling a signal?

40.What are the characteristics of a narrow band lowpass filter?

41.What are types of filter banks?

42.Draw the structure of an analysis filter bank.

43.Draw the structure of a synthesis filter bank.

44.Draw the structure of a two-channel quadrature mirror filter bank.

45.What are the sections of a two channel QMF bank?

46. Explain the design procedures of narrow band filters.

47. What is aliasing? Explain the aliasing effect in digital filters with suitable waveforms. Mention the methods of eliminating the aliasing effect.

48. Write short notes on Sampling & Sampling Rate alteration.

49. Explain the different interpolation techniques with suitable examples.

50. Explain the principle of sampling rate conversion for multirate signal processing.

51. What is Nyquist sampling theorem? Explain Decimation by a factor of 'D' and Interpolation by a factor of 'L' with suitable examples.

52. Explain in detail the poly phase structures for decimation and interpolation filters.

53.Explain the design of narrow band filters in detail.

54. With necessary equations and diagrams, discuss about the interpolation and decimation in

multirate signal processing.

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter '11' page no's(750- 782)
2. Digital signal processing by P.Ramesh babu Refer Chapter '8' page no's(8.1-8.51)

UNIT-VIII:FINITE WORD LENGTH EFECTS:

1. What do you understand by a fixed-point number?
2. What is the objective of spectrum estimation?
3. What is meant by block floating point representation? What are its advantages?
4. What are the advantages of floating point arithmetic?
5. How the multiplication & addition are carried out in floating point arithmetic?
6. What do you understand by input quantization error?
7. What is the relationship between truncation error e and the bits b for representing a decimal into binary?
8. What is meant rounding? Discuss its effect on all types of number representation?
9. What is the effect of quantization on pole location?
10. What is meant by quantization step size?
11. How would you relate the steady-state noise power due to quantization and the b bits representing the binary sequence?
12. What is overflow oscillation?
13. What are the methods used to prevent overflow?
14. What are the two kinds of limit cycle behavior in DSP?
15. Determine "dead band" of the filter.
16. Explain briefly the need for scaling in the digital filter implementation.

17. Under what conditions a finite duration sequence h(n) will yield constant group delay in its frequency response characteristics and not the phase delay?

18. Explain the operation of prevent overflow

19. Explain the limit cycles and overflow oscillations.

20. Explain about the round –off noise in IIR filters

21. Explain about the round –off noise in FIR filters

22. Explain about the computational output round off noise

23. Explain about the methods to prevents overflow

24. Explain about the trade –off between round off and overflow noise

25. Explain about the dead band effects.

26. List the applications of multirate signal processing.

27.State the Nyquist sampling theorem.

28.Distinguish Upsampling & Downsampling.

29.What is aliasing? Mention the cause producing aliasing effect.

30.Define decimation and interpolation.

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45.What are the sections of a two channel QMF bank?

TEXT BOOKS:

1. Digital Signal Processing, Principles, Algorithms, and Applications: John G. Proakis, Dimitris G. Manolakis. Pearson Education / PHI. 2007. Refer Chapter '9' page no's(601- 640)
2. Digital signal processing by P.Ramesh babu Refer Chapter '7' page no's(7.1-7.69)