R.M.D ENGINEERING COLLEGE
DEPARTMENT OF ECE
QUESTION BANK
DIGITAL SIGNAL PROCESSING

BRANCH/SEM/SEC:CSE/IV/A& B

UNIT I

SIGNALS AND SYSTEMS

Part – A

1. What do you understand by the terms : signal and signal processing
2. Determine which of the following signals are periodic and compute their fundamental period (AU DEC 07)
   a) sin√2 Лt b)sin20 Лt+sin5Лt
3. What are energy and power signals? (MU Oct. 96)
4. State the convolution property of Z transform (AU DEC 06)
5. Test the following systems for time invariance: (DEC 03)
   a) $y(n)=n\,x^2\,(n)$ b)$a^{x(n)}$
6. Define symmetric and antisymmetric signals. How do you prevent alaising while sampling a CT signal? (AU MAY 07)(EC 333, May '07)
7. What are the properties of region of convergence(ROC) ?(AU MAY 07)
8. Differentiate between recursive and non recursive difference equations (AU APR 05)
9. What are the different types of signal representation?
10. Define correlation (AU DEC 04)
11. what is the causality condition for LTI systems? (AU DEC 04)
12. define linear convolution of two DT signals (AU APR 04)
13. Define system function and stability of DT system (AU APR 04)
14. Define the following (a) System (b) Discrete-time system
15. What are the classifications of discrete-time system?
16. What is the property of shift-invariant system?
17. Define (a) Static system (b) Dynamic system? (AU DEC 03)
18. define cumulative and associative law of convolution (AU DEC 03)
19. Define a stable and causal system
20. What is the necessary and sufficient condition on the impulse response for stability? (MU APR.96)
21. What do you understand by linear convolution? (MU APR. 2000)
22. What are the properties of convolution? (AU IT Dec. 03)
23. State Parseval's energy theorem for discrete-time aperiodic signals(AU DEC 04)
24. Define DTFT pair (MU Apr. 99)
25. What is aliasing effect? (AU MAY 07) (EC 333 DEC 03)
26. State sampling theorem

27. What is an anti-aliasing filter?

28. What is the necessary and sufficient condition on the impulse response for stability? (EC 333, May '07)

22. State the condition for a digital filter to be causal and stable

Part-B

1. a) Compute the convolution y(n) of the signals (AU DEC 07)

$x(n)= a^n \quad -3 \le n \le 5$
$\quad 0 \quad$ elsewhere and
$h(n)= 1 \quad 0 \le n \le 4$
$\quad 0 \quad$ elsewhere

b) A discrete time system can be static or dynamic, linear or non-linear, Time variant or time invariant, causal or non causal, stable or unstable. Examine the following system with respect to the properties also (AU DEC 07)

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

2. a) Determine the response of the causal system
y(n)-y(n-1)=x(n)+x(n-1) to inputs x(n)=u(n) and $x(n)=2^{-n}u(n)$.Test its stability
b) Determine the IZT of $X(Z)=1/(1-Z^{-1})(1-Z^{-1})^2$ (AU DEC 07)

Determine whether each of the following systems defined below is (i) casual (ii) linear (iii) dynamic (iv) time invariant (v) stable

(a) $y(n) = \log_{10}[\{x(n)\}]$
(b) y(n) = x(-n-2)
(c) y(n) = cosh[nx(n) + x(n-1)]

3. Compute the convolution of the following signals

x(n) = {1,0,2,5,4} h(n) = {1,-1,1,-1}
↑ ↑

h(n) = {1,0,1} x(n) = {1,-2,-2,3,4}
↑ ↑

4. Find the convolution of the two signals

$x(n) = 3^n u(-n)$; $h(n) = (1/3)^n u(n-2)$
$x(n) = (1/3)^{-n} u(-n-1)$; h(n) = u(n-1)
x(n) = u(n) –u(n-5); h(n) = 2[u(n) – u(n-3)]

5. Find the discrete-time Fourier transform of the following
    $x(n) = 2^{-2n}$ for all n
    $x(n) = 2^n u(-n)$
    x(n) = n [1/2] (n)

6. Determine and sketch the magnitude and phase response of the following systems
    (a) y(n) = 1/3 [x(n) + x(n-1) + x(n-2)]
    (b) y(n) = ½[x(n) – x(n-1)]
    (c) y(n) - 1/2y(n-1)=x(n)

7. a) Determine the impulse response of the filter defined by y(n)=x(n)+by(n-1)
   b) A system has unit sample response h(n) given by
   h(n)=-1/δ(n+1)+1/2δ(n)-1-1/4 δ(n-1). Is the system BIBO stable? Is the filter causal? Justify your answer (DEC 2003)

8. Determine the Fourier transform of the following two signals(CS 331 DEC 2003)
    a) a^n u(n) for a<1
    b) cos ωn u(n)

9. Check whether the following systems are linear or not (AU APR 05)
    a) $y(n)=x^2(n)$ b) y(n)=n x(n)

10. For each impulse response listed below, dtermine if the corresponding system is
    i) causal ii) stable (AU MAY 07)
        1) 2^n u(-n)
        2) sin nЛ/2 (AU DEC 04)
        3) δ(n)+sin nЛ
        4) e^{2n} u(n-1)

11. Explain with suitable block diagram in detail about the analog to digital conversion and to reconstruct the analog signal (AU DEC 07)

12. Find the cross correlation of two sequences
    x(n)={1,2,1,1}   y(n)={1,1,2,1} (AU DEC 04)

13. Determine whether the following systems are linear , time invariant
        1) y(n)=A x(n)+B
        2) y(n)=x(2n)
    Find the convolution of the following sequences: (AU DEC 04)
        1) x(n)=u(n) h(n)=u(n-3)
        2) x(n)={1,2,-1,1} h(n)={1,0,1,1}

UNIT II

FAST FOURIER TRANSFORMS

1) THE DISCRETE FOURIER TRANSFORM

PART A

1. Find the N-point DFT of a sequence x(n) ={1 ,1, 2, 2}
2. Determine the circular convolution of the sequence x1(n)={1,2,3,1} and x2(n)={4,3,2,1} (AU DEC 07)
3. Draw the basic butterfly diagram for radix 2 DIT-FFT and DIF-FFT(AU DEC 07)
4. Determine the DTFT of the sequence $x(n)=a^{n} u(n)$ for a<1 (AU DEC 06)
5. Is the DFT of the finite length sequence periodic? If so state the reason
   (AU DEC 05)
6. Find the N-point IDFT of a sequence X(k) ={1 ,0 ,0 ,0} (Oct 98)
7. what do you mean by 'in place' computation of FFT? (AU DEC 05)
8. What is zero padding? What are its uses? (AU DEC 04)
9. List out the properties of DFT (MU Oct 95,98,Apr 2000)
10. Compute the DFT of $x(n)=\partial(n-n_0)$
11. Find the DFT of the sequence of $x(n)= \cos (n\prod/4)$ for $0\leq n\geq 3$ (MU Oct 98)
12. Compute the DFT of the sequence whose values for one period is given by x(n)={1,1,-2,-2}. (AU Nov 06,MU Apr 99)
13. Find the IDFT of Y(k)={1,0,1,0} (MU Oct 98)
14. What is zero padding? What are its uses?
15. Define discrete Fourier series.
16. Define circular convolution
17. Distinguish between linear convolution and Circular Convolution.
    (MU Oct 96,Oct 97,Oct 98)
18. Obtain the circular convolution of the following sequences x(n)={1, 2, 1} and h(n)={1, -2, 2}
19. Distinguish between DFT and DTFT (AU APR 04)
20. Write the analysis and synthesis equation of DFT (AU DEC 03)
21. Assume two finite duration sequences $x_1(n)$ and $x_2(n)$ are linearly combined. What is the DFT of x3(n)?(x3(n)=Ax1(n)+Bx2(n)) (MU Oct 95)
22. If X(k) is a DFT of a sequence x(n) then what is the DFT of real part of x(n)?
23. Calculate the DFT of a sequence x(n)=(1/4)^n for N=16 (MU Oct 97)
24. State and prove time shifting property of DFT (MU Oct 98)
25. Establish the relation between DFT and Z transform (MU Oct 98,Apr 99,Oct 00)
26. What do you understand by Periodic convolution? (MU Oct 00)
27. How the circular convolution is obtained using concentric circle method?
    (MU Apr 98)
28. State the circular time shifting and circular frequency shifting properties of DFT
29. State and prove Parseval's theorem
30. Find the circular convolution of the two sequences using matrix method
    X1(n)={1, 2, 3, 4} and x2(n)={1, 1, 1, 1}

31. State the time reversal property of DFT
32. If the DFT of x(n) is X(k) then what is the DFT of x*(n)?
33. State circular convolution and circular correlation properties of DFT
34. Find the circular convolution of the following two sequences using concentric circle method
    $x_1(n)=\{1, 2, 3, 4\}$ and $x_2(n)=\{1, 1, 1, 1\}$
35. The first five coefficients of X(K)={1, 0.2+5j, 2+3j, 2 ,5 }Find the remaining coefficients

PART B

1. 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) $x(n)=2^n$
(d) x(n)=sin(n∏/2)

2. 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}

3. 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}

4. Find the circular convolution of the following using matrix method and concentric circle method
(a) $x_1(n)=\{1,-1,2,3\}$; $x_2(n)=\{1,1,1\}$;
(b) $x_1(n)=\{2,3,-1,2\}$; $x_2(n)=\{-1,2,-1,2\}$;
(c) $x_1(n)=\sin n∏/2$; $x_2(n)=3^n$ 0≤n≥7

5.Calculate the DFT of the sequence x(n)={1,1,-2,-2}
Determine the response of the LTI system by radix2 DIT-FFT? (AU Nov 06).
If the impulse response of a LTI system is h(n)=(1,2,3,-1)
⇧

6. Determine the impulse response for the cascade of two LTI systems having impulse responses $h_1(n)=(1/2)$^n* u(n),$h_2(n)=(1/4)$^n*u(n) (AU May 07)

7. Determine the circular convolution of the two sequences $x_1(n)=\{1, 2, 3, 4\}$ $x_2(n)=\{1, 1, 1, 1\}$ and prove that it is equal to the linear convolution of the same.

8. Find the output sequence y(n)if h(n)={1,1,1,1} and x(n)={1,2,3,1} using circular convolution (AU APR 04)

9. State and prove the following properties of DFT (AU DEC 03)
   1) Cirular convolution 2) Parseval's relation
   2) Find the circular convolution of x1(n)={1,2,3,4} x2(n)={4,3,2,1}

2) FAST FOURIER TRANSFORM

PART A

1. Why FFT is needed? (AU DEC 03) (MU Oct 95,Apr 98)
2. What is FFT? (AU DEC 06)
3. Obtain the block diagram representation of the FIR filter (AU DEC 06)
4. Calculate the number of multiplications needed in the calculation of DFT and FFT with 64 point sequence. (MU Oct 97, 98).
5. What is the main advantage of FFT?
6. What is FFT? (AU Nov 06)
7. How many multiplications and additions are required to compute N-point DFT using radix 2 FFT? (AU DEC 04)
8. Draw the direct form realization of FIR system (AU DEC 04)
9. What is decimation-in-time algorithm? (MU Oct 95).
10. What do you mean by 'in place' computation in DIT-FFT algorithm? (AU APR 04)
11. What is decimation-in-frequency algorithm? (MU Oct 95,Apr 98).
12. Mention the advantage of direct and cascade structures (AU APR 04)
13. Draw the direct form realization of the system y(n)=0.5x(n)+0.9y(n-1) (AU APR 05)
14. Draw the flow graph of a two point DFT for a DIT decomposition.
15. Draw the basic butterfly diagram for DIT and DIF algorithm. (AU 07).
16. How do we can calculate IDFT using FFT algorithm?
17. What are the applications of FFT algorithms?
18. Find the DFT of sequence x(n)={1,2,3,0} using DIT-FFT algorithms
19. Find the DFT of sequence x(n)={1,1, 1, 1} using DIF-FFT algorithms (AU DEC 04)

PART B

1. Compute an 8-point DFT of the following sequences using DIT and DIF algorithms
   (a)x(n)={1,-1,1,-1,0,0,0,0}
   (b)x(n)={1,1,1,1,1,1,1,1} (AU APR 05)
   (c)x(n)={0.5,0,0.5,0,0.5,0,0.5,0}
   (d)x(n)={1,2,3,2,1,2,3,2}
   (e)x(n)={0,0,1,1,1,1,0,0} (AU APR 04)

2. Compute the 8 point DFT of the sequence $x(n)=\{0.5, 0.5 ,0.5,0.5,0,0,0,0\}$ using radix 2 DIF and DIT algorithm (AU DEC 07)

3. a) Discuss the properties of DFT
   b) Discuss the use of FFT algorithm in linear filtering (AU DEC 07)

4. How do you linear filtering by FFT using save-add method (AU DEC 06)

5. Compute the IDFT of the following sequences using (a)DIT algorithm (b)DIF algorithms
   (a)$X(k)=\{1,1+j,1-j2,1,0,1+j2,1+j\}$
   (b)$X(k)=\{12,0,0,0,4,0,0,0\}$
   (c)$X(k)=\{5,0,1-j,0,1,0,1+j,0\}$
   (d)$X(k)=\{8,1+j2,1-j,0,1,0,1+j,1-j2\}$
   (e)$X(k)=\{16,1-j4.4142,0,1+j0.4142,0,1-j0.4142,0,1+j4.4142\}$

6. Derive the equation for DIT algorithm of FFT.
   How do you do linear filtering by FFT using Save Add method? (AU Nov 06)

7. a) From first principles obtain the signal flow graph for computing 8 point DFT using radix 2 DIT-FFT algorithm.

   b) Using the above signal flow graph compute DFT of $x(n)=\cos(n*\text{Л})/4$,$0<=n<=7$ (AU May 07).

8. Draw the butterfly diagram using 8 pt DIT-FFT for the following sequences
   $x(n)=\{1,0,0,0,0,0,0,0\}$ (AU May 07).

9. a) From first principles obtain the signal flow graph for computing 8 point DFT using radix 2 DIF-FFT algorithm.
   b) Using the above signal flow graph compute DFT of $x(n)=\cos(n*\text{Л})/4$,$0<=n<=7$

10. State and prove circular time shift and circular frequency shift properties of DFT

11. State and prove circular convolution and circular conjugate properties of DFT

12. Explain the use of FFT algorithms in linear filtering and correlation

13. Determine the direct form realization of the following system
    $y(n)=-0.1y(n-1)+0.72y(n-2)+0.7x(n)-0.252x(n-2)$ (AU APR 05)

14. Determine the cascade and parallel form realization of the following system
$y(n)=-0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)$
    Expalin in detail about the round off errors in digital filters (AU DEC 04)

UNIT-III

IIR FILTER DESIGN

PART-A

1. Distinguish between Butterworth and Chebyshev filter
2. What is prewarping? (AU DEC 03)
3. Distinguish between FIR and IIR filters (AU DEC 07)
4. Give any two properties of Butterworth and chebyshev filters (AU DEC 06)
5. Give the bilinear transformation (AU DEC 03)
6. Determine the order of the analog butterworth filter that has a -2 dB pass band attenuation at a frequency of 20 rad/sec and atleast -10 dB stop band attenuation at 30 rad/sec (AU DEC 07)
7. By impulse invariant method obtain the digital filter transfer function and differential equation of the analog filter H(S)=1/S+1 (AU DEC 07)
8. Give the expression for location of poles of normalized butterworth filter (EC 333, May ‘07)
9. What are the parameters(specifications) of a chebyshev filter (EC 333, May ‘07)
10. Why impulse invariance method is not preferred in the design of IIR filter other than low pass filter?
11. What are the advantages and disadvantages of bilinear transformation?(AU DEC 04)
12. Write down the transfer function of the first order butterworth filter having low pass behavior (AU APR 05)

13. What is warping effect? What is its effect on magnitude and phase response?
14. Find the digital filter transfer function H(Z) by using impulse invariance method for the analog transfer function H(S)= 1/S+2 (MAY AU ’07)
15. Find the digital filter transfer function H(Z) by using bilinear transformation method for the analog transfer function H(S)= 1/S+3
16. Give the equation for converting a normalized LPF into a BPF with cutoff frequencies Ω_l and Ω_u
17. Give the magnitude function of Butterworth filter. What is the effect of varying order of N on magnitude and phase response?
18. Give any two properties of Butterworth low pass filters. (MU NOV 06).
19. What are the properties of Chebyshev filter? (AU NOV 06).
20. Give the equation for the order of N and cut off frequency Ω_c of Butterworth filter.
21. Give the Chebyshev filter transfer function and its magnitude response.
22. Distinguish between the frequency response of Chebyshev Type I filter for N odd and N even.
23. Distinguish between the frequency response of Chebyshev Type I & Type II filter.
24. Give the Butterworth filter transfer function and its magnitude characteristics for different order of filters.
25. Give the equations for the order N, major, minor and axis of an ellipse in case of Chebyshev filter.
26. What are the parameters that can be obtained from the Chebyshev filter specification? (AU MAY 07).

27. Give the expression for the location of poles and zeros of a Chebyshev Type II filter.
28. Give the expression for location of poles for a Chebyshev Type I filter. (AU MAY 07)
29. Distinguish between Butterworth and Chebyshev Type I filter.
30. How one can design Digital filters from Analog filters.
31. Mention any two procedures for digitizing the transfer function of an analog filter. (AU APR 04)
32. What are properties that are maintained same in the transfer of analog filter into a digital filter.
33. What is the mapping procedure between s-plane and z-plane in the method of mapping of differentials? What is its characteristics?
34. What is mean by Impulse invariant method of designing IIR filter?
35. What are the different types of structures for the realization of IIR systems?
36. Write short notes on prewarping.
37. What are the advantages and disadvantages of Bilinear transformation?
38. What is warping effect? What is its effect on magnitude and phase response?
39. What is Bilinear Transformation?
40. How many numbers of additions, multiplications and memory locations are required to realize a system H(z) having M zeros and N poles in direct form-I and direct form –II realization?
41. Define signal flow graph.
42. What is the transposition theorem and transposed structure?
43. Draw the parallel form structure of IIR filter.
44. Give the transposed direct form –II structure of IIR second order system.
45. What are the different types of filters based on impulse response? (AU 07)
46. What is the most general form of IIR filter?

PART B

1. a) Derive bilinear transformation for an analog filter with system function H(S)=b/S+a (AU DEC 07)

   b) Design a single pole low pass digital IIR filter with-3 Db bandwidth of 0.2Л by using bilinear transformation

2. a) Obtain the direct form I, Direct form II, cascade and parallel realization for the following

   Systems

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

   b) Discuss the limitation of designing an IIR filetr using impulse invariant method (AU DEC 07)

3. Determine H(Z) for a Butterworth filter satisfying the following specifications:

   $$0.8 \le |H(e^{j\omega}| \le 1, \text{ for } 0 \le \omega \le \pi/4$$
   $$|H(e^{j\omega}| \le 0.2, \text{ for } \pi/2 \le \omega \le \pi$$

   Assume T= 0.1 sec. Apply bilinear transformation method (AU MAY 07)

4.Determine digital Butterworth filter satisfying the following specifications:

$$0.707 \le |H(e^{j\omega}| \le 1, \text{ for } 0 \le \omega \le \pi/2$$
$$|H(e^{j\omega}| \le 0.2, \text{ for} 3\pi/4 \le \omega \le \pi$$

Assume T= 1 sec. Apply bilinear transformation method. Realize the filter in mose convenient form (AU DEC 06)

5. Design a Chebyshev lowpass filter with the specifications α_p=1 dB ripple in the pass band $0\leq\omega\leq0.2\pi$, α_s=15 dB ripple in the stop band $0.3\ \pi\leq\omega\leq\pi$ using impulse invariance method(AU DEC 06)

6. Design a Butterworth high pass filter satisfying the following specifications.
   α_p =1 dB; α_s=15 dB
   Ω_p =0.4Π; Ω_s =0.2Π

7. Design a Butterworth low pass filter satisfying the following specifications. (AU DEC 04)
   f_p=0.10 Hz;α_p=0.5 dB
   f_s=0.15 HZ;α_s=15 dB:F=1Hz.

8. Design (a) a Butterworth and (b) a Chebyshev analog high pass filter that will pass all radian frequencies greater than 200 rad/sec with no more that 2 dB attuenuation and have a stopband attenuation of greater than 20 dB for all Ω less than 100 rad/sec.

9. Design a digital filter equivalent to this using impulse invariant method
   $H(S)=10/S^2+7S+10$ (AU DEC 03)(AU DEC 04)

10. Use impulse invariance to obtain H(Z) if T= 1 sec and H(s) is
    $1/(s^3 +3s^2 +4s+1)$
    $1/(s^2+\surd2\ s +1)$

11. Use bilinear transformation method to obtain H(Z) if T= 1 sec and H(s) is
    1/(s+1)(S+2) (AU DEC 03)
    $1/(s^2+\surd2\ s +1)$

12. Briefly explain about bilinear transformation of digital filter design(AU APR 05)

13. Use bilinear transform to design a butterworth LPF with 3 dB cutoff frequeny of 0.2Π (AU APR 04)

14. Compare bilinear transformation and impulse invariant mapping

15. a) Design a chebyshev filter with a maxmimum pass band attenuation of 2.5 Db; at Ωp=20 rad/sec and the stop band attenuation of 30 Db at Ωs=50 rad/sec.
    b)Realize the system given by difference equation
    y(n)=-0.1 y(n-1)+0.72y(n-2)+0.7x(n)-0.25x(n-2) in parallel form
    (EC 333 DEC ‘07)

UNIT IV

FIR FILTER DESIGN

PART A

1. What are the desirable and undesirable features of FIR filter?
2. Discuss the stability of the FIR filters (AU APR 04) (AU DEC 03)
3. What are the main advantages of FIR over IIR (AU APR 04)
4. What is the condition satisfied by Linear phase FIR filter? (DEC 04) (EC 333 MAY 07)
5. What are the design techniques of designing FIR filters?
6. What condition on the FIR sequence h(n) are to be imposed in order that this filter can be called a Linear phase filter? (AU 07)
7. State the condition for a digital filter to be a causal and stable. (AU 06)
8. What is Gibbs phenomenon? (AU DEC 04) (AU DEC 07)
9. Show that the filter with h(n)={-1, 0, 1} is a linear phase filter
10. Explain the procedure for designing FIR filters using windows. (MU 02)
11. What are desirable characteristics of windows?
12. What is the principle of designing FIR filters using windows?
13. What is a window and why it is necessary?
14. Draw the frequency response of N point rectangular window. (MU 03)
15. Give the equation specifying Hanning and Blackman windows.
16. Give the expression for the frequency response of
17. Draw the frequency response of N point Bartlett window
18. Draw the frequency response of N point Blackman window
19. Draw the frequency response of N point Hanning window. (AU DEC 03)
20. What is the necessary and sufficient condition for linear phase characteristics in FIR filter. (MU Nov 03)
21. Give the equation specifying Kaiser window.
22. Compare rectangular and hanning window functions
23. Briefly explain the frequency sampling method of filter design
24. Compare frequency sampling and windowing method of filter design

PART-B

1. Use window method with a Hamming window to design a 13-tap differentiator (N=13). (AU ‘07)

2. i) Prove that FIR filter has linear phase if the unit impulse responsesatisfies the condition h(n)=h(N-1-n), n=0,1,……M-1. Also discuss symmetric and antisymmetric cases of FIR filter (AU DEC 07)

3. What are the issues in designing FIR filter using window method?(AU APR 04, DEC 03)

4. ii) Explain the need for the use of window sequences in the design of FIR filter. Describe the window sequences generally used and compare their properties

5. Derive the frequency response of a linear phase FIR filter when impulse responses symmetric & order N is EVEN and mention its applications

6. i) Explain the type I design of FIR filter using frequency sampling method
   ii) A low pass filter has the desired response as given below
   Hd($e^{j\omega}$)= e –j3ω, 0≤ω≤Л/2
   0 Л/2≤ω≤Л
   Determine the filter coefficients h(n) for M=7 using frequency sampling technique (AU DEC 07)

7. i) Derive the frequency response of a linear phase FIR filter when impulse responses antisymmetric & order N is odd
   ii) Explain design of FIR filter by frequency sampling technique (AU MAY 07)

7. Design an approximation to an ideal bandpass filter with magnitude response
   H($e^{j\omega}$) = 1 ; ∏/4≤|ω|≤3∏/4
   0 ; otherwise
   Take N=11. (AU DEC 04)

8. Design a 15-tap linear phase filter to the following discrete frequency response (N=15) using frequency sampling method (MU 03)
   H(k) = 1 $0 \le k \le 4$
   = 0.5 k=5
   = 0.25 k=6
   = 0.1 k=7
   = 0 elsewhere

9. Design an ideal band pass digital FIR filter with desired frequency response
   H($e^{j\omega}$)= 1 for $0.25\pi \le |\omega| \le 0.75\pi$
   0 for $|\omega| \le 0.25\pi$ and $0.75\pi \le |\omega| \le \pi$
   by using rectangular window function of length N=11. (AU DEC 07)

10. Design an Ideal Hilbert transformer using hanning window and Blackman window for N=11. Plot the frequency response in both Cases

11. a) How is the design of linear phase FIR filter done by frequency sampling method? Explain.
    b) Determine the coefficients of a linear phase FIR filter of length N=15 which has Symmetric unit sample response and a frequency response that satisfies the following conditions

    $H_r(2\pi k/15)$ = 1 for k=0,1,2,3

0 for k=4
0 for k=5,6,7

12. An FIR filter is given by the difference equation
y(n)=2x(n)+4/5 x(n-1)+3/2 x(n-2)+2/3 x(n-3) Determine its lattice form(EC 333 DEC 07)

13. Using a rectangular window technique design a low pass filter with pass band gain of unity cut off frequency of 1000 Hz and working at a sampling frequency of 5 KHz. The length of the impulse response should be 7.(EC 333 DEC 07)

16. Design an Ideal Hilbert transformer using rectangular window and Black man window for N=11. Plot the frequency response in both Cases (EC 333 DEC '07)

9. 17. Design an approximation to an ideal lowpass filter with magnitude response
$H(e^{j\omega}) = 1$; $0\leq|\omega|\leq\prod/4$
0 ; otherwise
Take N=11.Use hanning and hamming window (AU DEC 04)

UNIT V

FINITE WORD LENGTH EFFECTS

PART –A

1. What do you understand by a fixed point number? (MU Oct'95)
2. Express the fraction 7/8 and -7/8 in sign magnitude, 2's complement and 1's complement (AU DEC 06)
3. What are the quantization errors due to finite word length registers in digital filters? (AU DEC 06)
4. What are the different quantization methods? (AU DEC 07)
5. What are the different types of fixed point number representation?
6. What do you understand by sign-magnitude representation?
7. What do you understand by 2's complement representation?
8. Write an account on floating point arithmetic? (MU Apr 2000)
9. What is meant by block floating point representation? What are its advantages?
10. what are advantages of floating point arithmetic?
11. Compare the fixed point and floating point arithmetic. (MU Oct'96)
12. What are the three quantization errors due to finite word length registers in digital filters? (MU Oct'98)
13. How the multiplication and addition are carried out in floating point arithmetic?
14. Brief on co-efficient inaccuracy.
15. What do you understand by input quantization error?
16. What is product quantization error?
17. What is meant by A/D conversion mode?
18. What is the effect of quantization on pole locations?

19. What are the assumptions made concerning the statistical independence of various noise sources that occur in realizing the filter? (M.U. Apr 96)
20. What is zero input limit cycle overflow oscillation (AU 07)
21. What is meant by limit cycle oscillations?(M.U Oct 97, 98, Apr 2000) (AU DEC 07)
29. Explain briefly the need for scaling digital filter implementation? (M.U Oct 98)(AU-DEC 07)
30. Why rounding is preferred than truncation in realizing digital filter? (M.U. Apr 00)
31. Define the deadband of the filter? (AU 06)
25. Determine the dead band of the filter with pole at 0.5 and the number of bits used for quantization is 4(including sign bit)
26. Draw the quantization noise model for a first order IIR system
27. What is meant by rounding? Draw the pdf of round off error
28. What is meant by truncation? Draw the pdf of round off error
29. What do you mean by quantization step size?
30. Find the quantization step size of the quantizer with 3 bits
31. Give the expression for signal to quantization noise ratio and calculate the improvement with an increase of 2 bits to the existing bit.
32. Express the following binary numbers in decimal

    A) $(100111.1110)_2$ (B) $(101110.1111)_2$ C $(10011.011)_2$

33.Why rounding is preferred to truncation in realizing digital filter? (EC 333, May '07)
34. List the different types of frequency domain coding (EC 333 MAY 07)
35. What is subband coding? (EC 333 MAY 07)

PART-B

1. Draw the quantization noise model for a second order system and explain $H(z)=1/(1-2r\cos\theta z^{-1}+r^2z^{-2})$ and find its steady state output noise variance (ECE AU' 05)

2. Consider the transfer function $H(z)=H_1(z)H_2(z)$ where $H_1(z)=1/(1-a_1z^{-1})$, $H_2(z)=1/(1-a_2z^{-2})$.Find the output round off noise power. Assume $a_1=0.5$ and $a_2=0.6$ and find out the output round off noise power. (ECE AU' 04)(EC 333 DEC 07)

3. Find the effect of coefficient quantiztion on pole locations of the given second order IIR system when it is realized in direct form –I and in cascade form. Assume a word length of 4-bits through truncation.
   $H(z)= 1/(1-0.9z^{-1}+0.2z^{-2})$ (AU' Nov 05)

4. Explain the characteristics of Limit cycle oscillations with respect to the system described by the differential equations.
   $y(n)=0.95y(n-1)+x(n)$ and
   determine the dead band of the filter (AU' Nov 04)
5. i) Describe the quantization errors that occur in rounding and truncation in two's complement
   ii) Draw a sample/hold circuit and explain its operation
   iii) What is a vocoder? Expalin with a block diagram (AU DEC 07)

6. Two first order low pass filter whose system functions are given below are connected in cascade. Determine the overall output noise power
   $H1(Z)=1/(1-0.9Z^{-1})$ $H2(Z)=1/(1-0.8Z^{-1})$ (AU DEC 07)

7. Consider a Butterworth lowpass filter whose transfer function is
$H(z)=0.05(1+z^{-1})^2 /(1-1.2z^{-1} +0.8 z^{-2})$.
Compute the pole positions in z-plane and calculate the scale factor S_o to prevent overflow in adder 1.

8. Express the following decimal numbers in binary form

A) 525 B) 152.1875 C) 225.3275

10. Express the decimal values 0.78125 and -0.1875 in

One's complement form

sign magnitude form

Two's complement form.

11. Express the decimal values -6/8 and 9/8 in (i) Sign magnitude form (ii) One's complement form (iii) Two's complement form

12. Study the limit cycle behavior of the following systems

    i. y(n) = 0.7y(n-1) + x (n)
    ii. y(n) = 0.65y(n-2) + 0.52y (n-1) + x (n)

13. For the system with system function H (z) =1+0.75z-1 / 1-0.4z-1 draw the signal flow graph
14. and find scale factor s0 to prevent overflow limit cycle oscillations
15. Derive the quantization input nose power and determine the signal to noise ratio of the system
16. Derive the truncation error and round off error noise power and compare both errors
17. Explain product quantization error and coefficient quantization error with examples
18. Derive the scaling factor So that prevents the overflow limit cycle oscillations in a second order IIR system.
19. The input to the system y(n)=0.999y(n-1)+x(n) is applied to an ADC. What is the power produced by the quantization noise at the output of the filter if the input is quantized to

1) 8 bits 2) 16 bits (EC 333 DEC 07)

19. Convert the following decimal numbers into binary: (EC 333 DEC 07)

1) $(20.675)_{10}$ 2) $(120.75)_{10}$

20. Find the steady state variance of the noise in the output due to quantization of input for the first order filter y(n)=ay(n-1)+x(n) (EC 333 DEC 07)

ANAND INSTITUTE OF HIGHER TECHNOLOGY
KAZHIPATTUR, CHENNAI –603 103

DEPARTMENT OF ECE

Date: 15-05-2009

PART-A QUESTIONS AND ANSWERS

Subject : Digital signal Processing **Sub Code : IT1252**

Staff Name: Robert Theivadas.J Class : VII Sem/CSE A&B

UNIT-1 - SIGNALS AND SYSTEMS

PART A

1. Determine which of the following sinusoids are periodic and compute their fundamental period

(a) Cos 0.01πn
(b) sin (π62n/10) **Nov/Dec 2008 CSE**

a) Cos 0.01 πn

Wo=0.01 π the fundamental frequency is multiply of π .Therefore the signal is periodic

Fundamental period

$N=2\pi\ [m/wo]$

$=2\pi(m/0.01\pi)$

Choose the smallest value of m that will make N an integer

M=0.1

$N=2\pi(0.1/0.01\pi)$

N=20

Fundamental period N=20

b) sin (π62n/10)

Wo=0.01 π the fundamental frequency is multiply of π .Therefore the signal is periodic

Fundamental period

$N=2\pi\ [m/wo]$

$=2\pi(m/(\pi62/10))$

Choose the smallest value of m that will make N an integer

M=31

$N=2\pi(310/62\pi)$

N=10

Fundamental period N=10

2. State sampling theorem **Nov/Dec 2008 CSE**

A band limited continuous time signal, with higher frequency f_{max} Hz can be uniquely recovered from its samples provided that the sampling rate $Fs>2f_{max}$ samples per second

3. State sampling theorem , and find Nyquist rate of the signal

x(t)=5 sin250 πt + 6cos300 πt **April/May2008 CSE**

A band limited continuous time signal, with higher frequency f_{max} Hz can be uniquely recovered from it's samples provided that the sampling rate $Fs>2f_{max}$ samples per second.

Nyquist rate

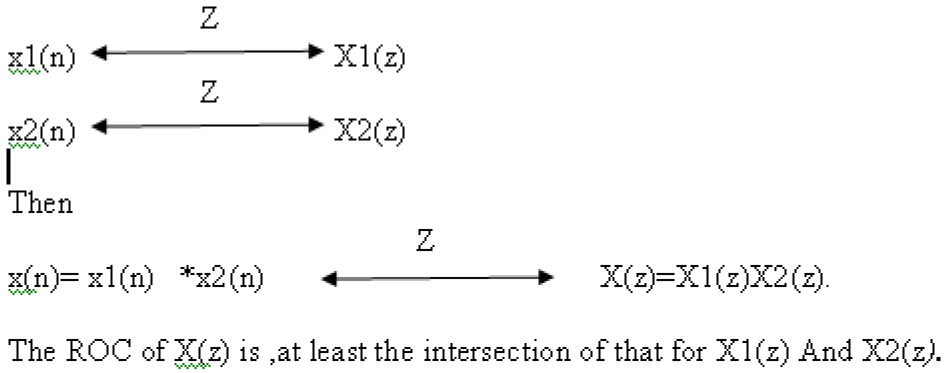
$x(t)=5\sin 250\pi t+ 6\cos 300\pi t$

Frequency present in the signals

$F1=125Hz$ $\quad F2=150Hz$
$Fmax=150Hz$
$Fs>2Fmax=300\ Hz$
The Nyquist rate is $FN= 300Hz$

4. State and prove convolution property of Z transform. April/May2008 CSE

Convolution Property (MAY 2006 ECESS)



$$x1(n) \xleftrightarrow{Z} X1(z)$$

$$x2(n) \xleftrightarrow{Z} X2(z)$$

Then

$$x(n)= x1(n) * x2(n) \xleftrightarrow{Z} X(z)=X1(z)X2(z).$$

The ROC of X(z) is ,at least the intersection of that for X1(z) And X2(z).

5. Determine which of the following signals are periodic and compute their fundamental period. Nov/Dec 2007 CSE

(a) sin √2пt
(b) sin 20пt + sin 5пt

(a) sin √2пt

wo=√2п .The Fundamental frequency is multiply of п.Therefore, the signal is Periodic .

Fundamental period

N=2п [m/wo]
= 2п [m/√2п]
m=√2
=2п [√2/√2п]
N=2

(b) sin 20пt + sin 5пt

wo=20п, 5п .The Fundamental frequency is multiply of п.Therefore, the signal is Periodic .

Fundamental period of signal sin 20пt

N1=2п [m/wo]
=2п [m/20п] m=1
=1/10

Fundamental period of signal sin 5пt

N2=2п [m/wo]
=2п [m/5п] m=1
=2/5
N1/N2=(1/10)/(2/5)
=1/4
4N1=N2

N= 4N1=N2
N=2/5

6. **Determine the circular convolution of the sequence x1(n)={1,2,3,1} and x2(n)={4,3,2,1}.** **Nov/Dec 2007 CSE**

Soln:

x1(n)={1,2,3,1}
x2(n)={4,3,2,1}.

$$\begin{vmatrix} x1(0) & x1(3) & x1(2) & x1(1) \\ x1(1) & x1(0) & x1(3) & x1(2) \\ x1(2) & x1(1) & x1(0) & x1(3) \\ x1(3) & x1(2) & x1(1) & x1(0) \end{vmatrix} \begin{vmatrix} x2(0) \\ x2(1) \\ x2(2) \\ x2(3) \end{vmatrix} = \begin{vmatrix} y(0) \\ y(1) \\ y(2) \\ y(3) \end{vmatrix}$$

$$\begin{vmatrix} 1 & 1 & 3 & 2 \\ 2 & 1 & 1 & 3 \\ 3 & 2 & 1 & 1 \\ 1 & 3 & 2 & 1 \end{vmatrix} \begin{vmatrix} 4 \\ 3 \\ 2 \\ 1 \end{vmatrix} = \begin{vmatrix} 15 \\ 16 \\ 21 \\ 15 \end{vmatrix}$$

Y(n)= $\{15,16,21,15\}$

7. **Define Z transform for x(n)=-na^{n} u(-n-1)** **April/May 2008 IT**

X(n) =-na^{n} u(-1-n)

X (z)=$\sum_{n=-\infty}^{\infty} x(n)\, z^{-n}$

=$\sum_{n=-\infty}^{\infty} -na^{n}\, u(-1-n) z^{-n}$ u(-n-1)=0for n>1

=$\sum_{n=-\infty}^{\infty} -na^{n} z^{-n}$
= -$\sum_{n=-\infty}^{\infty} na^{n} z^{-n}$
= -z d/dz X(z)
=z d/dz($\frac{1}{1-az^{-1}}$)= $az^{-1}/{(1-az^{-1})}^{2}$

8. **Find whether the signal y= n^{2} x(n) is linear** **April/May 2008 IT**

Y=n^2 x(n)

Y1(n)=T[x1(n)]= n^2 x1(n)

Y2(n)= T[x2(n)]= n^2 x2(n)

The weighted sum of input is

a1 T[x1(n)]+a2 T[x2(n)]=a1n^2 x1(n)+a2n^2 x2(n)-----------1

the output due to weighted sum of input is

y3(n)=T[a1X1(n)+a2X2(n)]

= a1n^2 x1(n)+a2n^2 x2(n)---------------------------------2

9. Is the system y(n)=ln[x(n)] is linear and time invariant? (MAY 2006 IT)

The system y(n)=ln[x(n)] is non-linear and time invariant

$a\ln x_1(n)+b\ln x_2(n) \neq \ln(ax_1(n)+bx_2(n))$ → Non-linear system

$\ln x(n)=\ln x(n-n_0)$ → Time invariant system

10. Write down the expression for discrete time unit impulse and unit step function. (APR 2005 IT).

Discrete time unit impulse function

$\delta(n) =1, n=0$

$=0, n\neq0$

Discrete time step impulse function.

$u(n) = 1$, for $n\geq0$

$= 0$ for $n<0$

11. List the properties of DT sinusoids. (NOV 2005 IT)

- DT sinusoid is periodic only if its frequency f is a rational number.
- DT sinusoid whose frequencies are separated by an integer multiple of 2π are identical.

12. Determine the response a system with y(n)=x(n-1) for the input signal

$x(n) = |n|$ for $-3\leq n\leq3$

$= 0$ otherwise (NOV 2005 IT)

$x(n)= \{3,2,1,0,1,2,3\}$

$\uparrow$

$y(n) = x(n-1) =\{3,2,1,0,1,2,3\}$

13. Define linear convolution of two DT signals. (APR 2004 IT)

y(n)=x(n)*h(n), * represent the convolution operator

y(n), x(n)&h(n), →Output, Input and response of the system respectively.

14. Define system function and stability of a DT system. (APR 2004 IT)

H(z)=Y(z)/X(z)

H(z),Y(z) & X(z)→z-transform of the system impulse, output and input respectively.

15. What is the causality condition for an LTI system? (NOV 2004 IT)

Conditions for the causality

h(n)=0 for n<0

16. What are the different methods of evaluating inverse z transform. (NOV 2004 IT)

- Long division method
- Partial fraction expansion method
- Residue method
- Convolution method

UNIT-II - FAST FOURIER TRANSFORMS

1. Find out the DFT of the signal X(n)= δ (n) Nov/Dec 2008 CSE

$$\delta(n) = \begin{cases} 1 \; for \; n = 0 \\ 0 \; for \; n \neq 0 \end{cases}$$

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

$$X(k) = \sum_{n=0}^{N-1} x(n)\, e^{-j2\pi nk/N} \qquad k = 0,1,2,3 \ldots\ldots.N-1$$

$$X(k) = \sum_{n=0}^{3} x(n)\, e^{-j2\pi nk/4} \qquad k = 0,1,2,3$$

$$X(k) = x(0) + x(1)e^{-\frac{jk\pi}{2}} + x(2)e^{-jk\pi} + x(3)e^{-j3k\pi/4} \qquad k = 0,1,2,3.$$

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

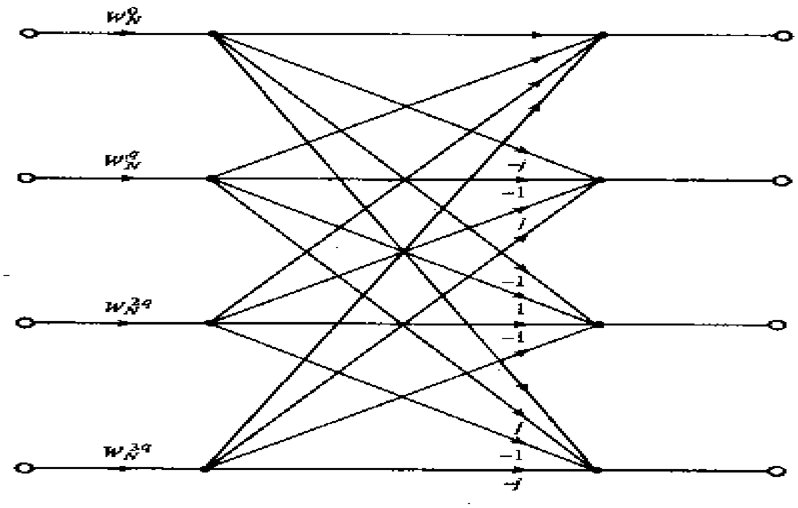
2. What is meant by bit reversal and in place commutation as applied to FFT?

Nov/Dec 2008
CSE

"Bit reversal" is just what it sounds like: reversing the bits in a binary word from left to write. Therefore the MSB's become LSB's and the LSB's become MSB's.The data ordering required by radix-2 FFT's turns out to be in "bit reversed" order, so bit-reversed indexes are used to combine FFT stages.

Input sample index	Binary Representation	Bit reversed binary	Bit reversal sample index
0	000	000	0
1	001	100	4
2	010	010	2
3	011	110	6
4	100	001	1
5	101	101	5
6	110	011	3
7	111	111	7

3. **Draw radix 4 butterfly structure for (DIT) FFT algorithm April/May2008 CSE**



4. Find DFT for {1,0,0,1}. April/May2008 CSE /April/May 2008 IT

$$x(k)= \sum_{n=0}^{N-1} x(n)e^{-j2\pi nk/N} \qquad K=0,1,2,3,\ldots N-1$$

$$x(k)= \sum_{n=0}^{3} x(n)e^{-j2\pi nk/N} \qquad K=0,1,2,3$$

$$N=4$$

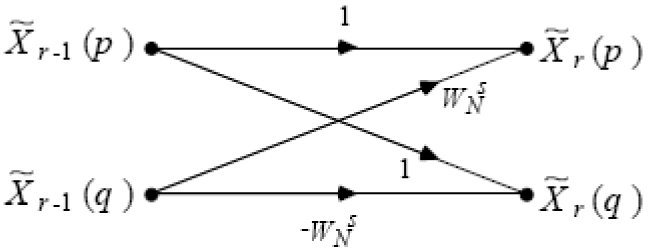
$$= x(0)+x(1)e^{-jk\pi/2}+x(2)e^{-jk\pi}+x(3)e^{-j3k\pi/2}$$

$$X(k)= 1+ e^{-j3k\pi/2} \qquad K=0,1,2,3$$

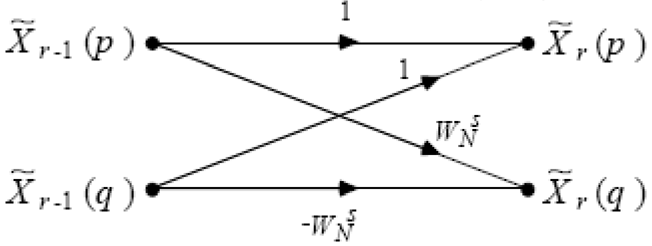
5. Draw the basic butterfly diagram for radix 2 DIT-FFT and DIF-FFT. Nov/Dec 2007 CSE

Butterfly Structure for DIT FFT **MAY 2006 ECESS &(NOV 2006 ITSS)**

The DIT structure can be expressed as a butterfly diagram



The DIF structure expressed as a butterfly diagram



6. What are the advantages of Bilinear mapping April/May 2008 IT

- Aliasing is avoided
- Mapping the S plane to the Z plane is one to one
- The closed left half of the S plane is mapped onto the unit disk of the Z plane

7. How may multiplication and addition is needed for radix-2 FFT? April/May 2008 IT

Number of complex addition is given by $N\log_2 N$

Number of complex multiplication is given by $N/2 \log_2 N$

8. Define DTFT pair? (May/June 2007)-ECE

The DTFT pairs are **(MAY 2006 IT)**

$X(k) = \quad x(n)e^{-j2\pi kn/N}$

$X(n) = \quad x(k)e^{j2\pi kn/N}$

9. Define Complex Conjugate of DFT property. (May/Jun 2007)-ECE

If $x(n)\underset{N}{\overset{DFT}{\leftrightarrow}}X(k)$ then

$X^*(n)\leftrightarrow(X^*(-k))_N = X^*(N- K)$

10.Differentiate between DIT and DIF FFT algorithms. (MAY 2006 IT)

S.No	DIT FFT algorithm	DIF FFT algorithm
1	Decimation in time FFT algorithm	Decimation in frequency FFT algorithm
2	Twiddle factor $k=(Nt/2^m)$	Twiddle factor $k=(Nt/2^{M-m+1})$

11.**Give any two properties of DFT (APR 2004 IT SS)**

Linearity : DFT [ax(n)+b y(n)]=a X(K)+bX(K)

Periodicity: x(n+N)=x(n) for all n
X(K+N)=X(K) for all n

12.What are the advantages of FFT algorithm over direct computation of DFT? (May/June 2007)-ECE

The complex multiplication in the FFT algorithm is reduced by (N/2) log2N times.
Processing speed is very high compared to the direct computation of DFT.

13. What is FFT? (Nov/Dec 2006)-ECE

The fast Fourier transform is an algorithm is used to calculate the DFT. It is based on fundamental principal of decomposing the computation of DFT of a sequence of the length N in to successively smaller discrete Fourier Transforms. The FFT algorithm provides speed increase factor when compared with direct computation of the DFT.

14.Determine the DIFT of a sequence $x(n) = a^n u(n)$ (Nov/Dec 2006)-ECE

$X(K) = \quad x(n)\, e^{j2\pi kn/N}$

The given sequence $x(n) = a^n u(n)$

$DTFT\{x(n)\} = \quad x(n)\, e^{j2\pi kn/N}$
$= \quad (a\, e^{j2\pi k/N})^n$

Where $a^n = 1\text{-}a^n/(1\text{-}a)$
$X(K) = (1 - a^N e^{j2\pi k})/\, (1\text{-}a e^{j2\pi k/N})$

15. What do you mean by in place computation in FFT. (APR 2005 IT)

FFT algorithms, for computing the DFT when the size N is a power of 2 and when it is a power of 4

16.Is the DFT is a finite length sequence periodic. Then state the reason (APR 2005 ITDSP)

DFT is a finite length sequence periodic.

$$X(e^{j\omega}) = \sum_{n=0}^{N-1} x(n)\, e^{-j\omega n}$$

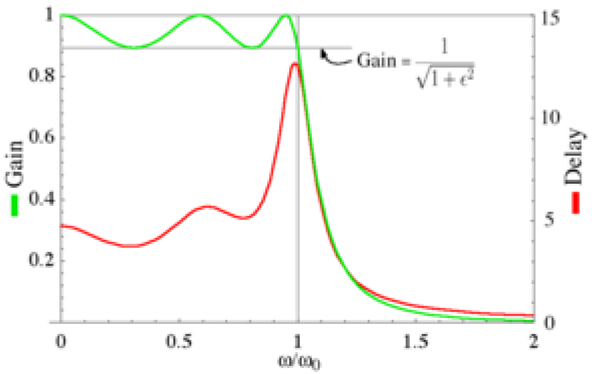
$X(e^{j\omega})$ is continuous & periodic in ω, with period 2π.

UNIT-III - IIR FILTER DESIGN

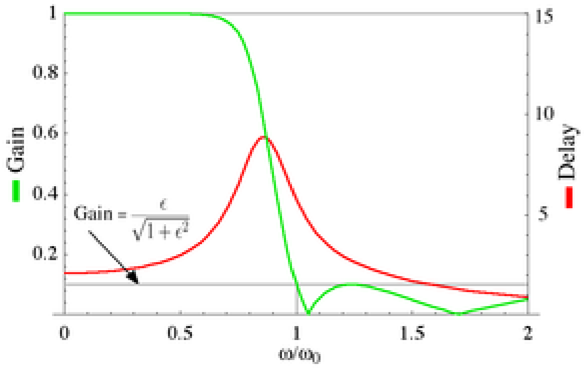
1. What are the requirements for converting a stable analog filter into a stable digital filter? Nov/Dec 2008 CSE

- The JΩ axis in the s plane should be map into the unit circle in the Z plane .thus there will be a direct relationship between the two frequency variables in the two domains
- The left half plane of the s plane should be map into the inside of the unit circle in the z –plane .thus the stable analog filter will be converted to a stable digital filter

2. Distinguish between the frequency response of chebyshev type I and Type II filter Nov/Dec 2008 CSE



Type I chebyshev filter



Type II chebyshev filter

Type I chebyshev filters are all pole filters that exhibit equirpple behavior in the pass band and monotonic in stop band .Type II chebyshev filters contain both poles and zeros and exhibits a monotonic behavior in the pass band and an equiripple behavior in the stop band

3. What is the need for prewraping in the design of IIR filter Nov/Dec 2008 CSE

The warping effect can be eliminated by prewarping the analog filter .This can be done by finding prewarping analog frequencies using the formula

$$\Omega = 2\tan^{-1}\Omega T/2$$

4.Write frequency translation for BPF from LPF April/May2008 CSE

Low pass with cut – off frequency Ω_C to band –pass with lower cut-off frequency Ω_1 and higher cut-off frequency Ω_2:

S ------------- Ω_C $(_{s2} + \Omega 1\ \Omega 2) / s\ (\Omega_2 - \Omega_1)$

The system function of the high pass filter is then

$$H(s) = H_p \{ \Omega_C (s^2 + \Omega 1\ \Omega 2) / s\ (\Omega_2 - \Omega_1)\}$$

5.Compare Butterworth, Chebyshev filters April/May2008 CSE

Butter Worth Filter	Chebyshev filters.
Magnitude response of Butterworth filter decreases monotonically, as frequency	Magnitude response of chebyshev filter exhibits ripple in pass band

→

increases from 0 ∞	
Poles on the butter worth lies on the circle	Poles of the chebyshev filter lies on the ellipse

6. Determine the order of the analog Butterworth filter that has a -2 db pass band attenuation at a frequency of 20 rad/sec and atleast -10 db stop band attenuation at 30 rad/sec.

Nov/Dec 2007CSE

α_p =2 dB; Ωp =20 rad/sec
α_s = 10 dB; Ωs = 30 rad/sec

$$N \geq \frac{\log\sqrt{10^{0.1\,\alpha s}-1/\,10^{0.1\,\alpha p}-1}}{\text{Log}\ \alpha_{s/}\ \alpha_p}$$

$$N \geq \frac{\log\sqrt{10-1/\,10^{0.2}-1}}{\text{Log}\ 30_{/}\ 20}$$

≥ 3.37

Rounding we get N=4

7. By Impulse Invariant method, obtain the digital filter transfer function and differential equation of the analog filter H(s)=1 / (s+1) Nov/Dec 2007 CSE

H(s) =1/(s+1)

Using partial fraction

H(s) =A/(s+1)
= 1/(s-(-1)

Using impulse invariance method

$H(z) = 1/1-e^{-T}z^{-1}$

AssumeT=1sec

$H(z)=1/1-e^{-1}z^{-1}$
$H(z)=1/1-0.3678z^{-1}$

8.Distinguish between FIR and IIR filters. Nov/Dec 2007 CSE

Sl.No	IIR	FIR
1	H(n) is infinite duration	H(n) is finite duration
2	Poles as well as zeros are present. Sometimes all pole filters are also designed.	These are all zero filters.
3	These filters use feedback from output. They are recursive filters.	These filters do not use feedback. They are nonrecursive.
4	Nonlinear phase response. Lin phase is obtained if $H(z) = \pm Z^{-1}H(Z^{-1})$	Linear phase response for $h(n) = \pm h(m-1-n)$
5	These filters are to be designed stability	These are inherently stable filters

6	Number of multiplication requirement is less.	More
7	More complexity of implementation	Less complexity of implementation
8	Less memory is required	More memory is requied
9	Design procedure is complication	Less complicated
10	Design methods: 1. Bilinear Transform 2. Impulse invariance.	Design methods: 1. Windowing 2. Frequency sampling
11	Can be used where sharp cutoff characteristics with minimum order are required	Used where linear phase characteristic is essential.

9.Define Parsevals relation **April/May 2008 IT**

If X1(n) and X2(n) are complex valued sequences ,then

$$\sum_{-\infty}^{\infty} x1(n)x2^{*}\ (n) = 1/2\prod j \oint_{c}\ X1(v)X2*(\frac{1}{v}*)v^{-1}\,dv$$

10.What are the advantages and disadvantages of bilinear transformation?
(May/June 2006)-ECE **Advantages:**

1. Many to one mapping.
2. linear frequency relationship between analog and its transformed digital frequency,

Disadvantage:

Aliasing

11.What is frequency warping? (MAY 2006 IT DSP)

The bilinear transform is a method of compressing the infinite, straight analog frequency axis to a finite one long enough to wrap around the unit circle only once. This is also sometimes called frequency warping. This introduces a distortion in the frequency. This is undone by pre-warping the critical frequencies of the analog filter (cutoff frequency, center frequency) such that when the analog filter is transformed into the digital filter, the designed digital filter will meet the desired specifications**.**

12. Give any two properties of Butterworth filter and chebyshev filter. (Nov/Dec 2006)

a. The magnitude response of the Butterworth filter decreases monotonically as the frequency increases (Ώ) from 0 to ∞.
b. The magnitude response of the Butterworth filter closely approximates the ideal response as the order N increases.
c. The poles on the Butterworth filter lies on the circle.
d. The magnitude response of the chebyshev type-I filter exhibits ripple in the pass band.
e. The poles of the Chebyshev type-I filter lies on an ellipse.

$S = (2/T)\ (Z-1)\ (Z+1)$

13.Find the transfer function for normalized Butterworth filter of order 1 by determining the pole values. **(MAY 2006 IT DSP)**

Poles = 2N
N=1
Poles = 2

14..Differentiate between recursive and non-recursive difference equations.
(APR 2005 ITDSP)

The FIR system is a non-recursive system, described by the difference equation M-1

$$y(n) = \sum_{k=0} b_k x(n-k)$$

The IIR system is a non-recursive system, described by the difference equation

$$y(n) = \sum_{k=0}^{N} b_k x(n-k) - \sum_{k=1}^{M} a_k y(n-k)$$

15.Find the order and poles of Butterworth LPF that has -3dB bandwidth of 500 Hz and an attenuation of -40 dB at 1000 Hz. (NOV 2005 ITDSP)

$\alpha_p = -3\text{dB}$ $\alpha_s = -40\text{dB}$ $\Omega_s = 1000*2\pi$ rad/sec $\Omega_p=500*2\pi$

The order of the filter $N \geq (\log(\lambda/\varepsilon))/(\log(\Omega_s/\Omega_p))$

$\lambda = (10^{0.1\alpha s}-1)^{1/2} = 99.995$

$\varepsilon = (10^{0.1\alpha p}-1)^{1/2} = 0.9976$

$N = (\log(99.995/0.9976))/(\log(2000\pi/1000\pi)) = 2/0.3 = 6.64$

$N \geq 6.64 = 7$

Poles=2N=14

16.What is impulse invariant mapping? What is its limitation? (Apr/May 2005)-ECE

The philosophy of this technique is to transform an analog prototype filter into an IIR discrete time filter whose impulse response [h(n)] is a sampled version of the analog filter's impulse response, multiplied by T.This procedure involves choosing the response of the digital filter as an equi-spaced sampled version of the analog filter.

17.Give the bilinear transformation. (Nov/Dec 2003)-ECE

The bilinear transformation method overcomes the effect of aliasing that is caused due to the analog frequency response containing components at or beyond the nyquist frequency. The bilinear transform is a method of compressing the infinite, straight analog frequency axis to a finite one long enough to wrap around the unit circle only once.

18.Mention advantages of direct form II and cascade structures. (APR 2004 ITDSP)

(i) The main advantage direct form-II structure realization is that the number of delay elements is reduced by half. Hence, the system complexity drastically reduces the number of memory elements .

(ii) Cascade structure realization, the system function is expressed as a product of several sub system functions. Each sum system in the cascade structure is realized in direct form-II. The order of each sub system may be two or three (depends) or more.

19. **What is prewarping? (Nov/Dec 2003)-ECE**

When bilinear transformation is applied, the discrete time frequency is related continuous time frequency as,

$$\Omega = 2\tan^{-1}\Omega T/2$$

This equation shows that frequency relationship is highly nonlinear. It is also called frequency warping. This effect can be nullified by applying prewarping. The specifications of equivalent analog filter are obtained by following relationship,

$$\Omega = 2/T \tan \omega/2$$

This is called prewarping relationship.

UNIT-IV - FIR FILTER DESIGN

1.What is gibb's Phenomenon. April/May2008 CSE

The oscillatory behavior of the approximation XN(W) to the function X(w) at a point of discontinuity of X(w) is called Gibb's Phenomenon

2.**Write procedure for designing FIR filter using windows. April/May2008 CSE**

1. Begin with the desired frequency response specification Hd(w)

2. Determine the corresponding unit sample response hd(n)
3. Indeed hd(n) is related to Hd(w) by the Fourier Transform relation.

3.**What are Gibbs oscillations?** **Nov/Dec 2007 CSE**

Oscillatory behavior observed when a square wave is reconstructed from finite number of harmonics.

The unit cell of the square wave is given by

$$Y(\nu) = \mathrm{rect}\left(\frac{\nu}{2\nu_c}\right)$$

Its Fourier series representation is

$$Y(\nu) = 2\nu_c \sum_{n=-\infty}^{\infty} \mathrm{sinc}(2n\nu_c)e^{-j2\pi n\nu}$$

4. **Explain briefly the need for scaling in the digital filter realization** **Nov/Dec 2007 CSE**

To prevent overflow, the signal level at certain points in the digital filters must be scaled so that no overflow occur in the adder

5. **What are the advantages of FIR filters?** **April/May 2008 IT**

1.FIR filter has exact linear phase
2.FIR filter always stable
3.FIR filter can be realized in both recursive and non recursive structure
4.Filters wit h any arbitrary magnitude response can be tackled using FIR sequency

6. Define Phase Dealy **April/May 2008 IT**

When the input signal X(n) is applied which has non zero response $\theta(w) = \arg\,[H(e^{jw})]$ the output signal y(n) experience a delay with respect to the input signal .Let the input signal be

X(n)=A $\cos[won + \emptyset]$, $+\infty > n > -\infty$

Where A= Maximum Amplitude of the signal
Wo=Frequency in radians
f=phase angle

Due to the delay in the system response ,the output signal lagging in phase $\theta(wo)$ but the frequency remain the same

Y(n)=$|H(e^{jw})|$A $\cos[won + \theta(wo) + \emptyset]$,

In This equation that the output is the time delayed signal and is more commonly known as phase delayed at w=wo $\tau_{p(wo)=-\theta(wo)/wo}$ Is called phase delay

7. State the advantages and disadvantages of FIR filter over IIR filter.

(MAY 2006 IT DSP) & (NOV 2004 ECEDSP)

Advantages of FIR filter over IIR filter

- It is a stable filter
- It exhibit linear phase, hence can be easily designed.
- It can be realized with recursive and non-recursive structures
- It is free of limit cycle oscillations when implemented on a finite word length digital system

Disadvantages of FIR filter over IIR filter

- Obtaining narrow transition band is more complex.
- Memory requirement is very high
- Execution time in processor implementation is very high.

8. List out the different forms of structural realization available for realizing a FIR system. (MAY 2006 IT DSP)

The different types of structures for realization of FIR system are

1.Direct form-I 2. Direct form-II

9. **What are the desirable and undesirable features of FIR Filters? (May/June 2006)-ECE**

The width of the main lobe should be small and it should contain as much of total energy as possible.The side lobes should decease in energy rapidly as w tends to π

10. Define Hanning and Blackman window functions. (May/June 2006)-ECE

The window function of a causal hanning window is given by

$W_{Hann}(n) = 0.5 - 0.5\cos 2\pi n/ (M-1), 0 \le n \le M-1$

0, Otherwise

The window function of non-causal Hanning window I s expressed by

$W_{Hann}(n) = 0.5 + 0.5\cos 2\pi n/ (M-1), 0 \le |n| \le (M-1)/2$

0, Otherwise

The width of the main lobe is approximately 8π/M and thee peak of the first side lobe is at -32dB**.**

The window function of a causal Blackman window is expressed by

$W_B(n) = 0.42 - 0.5 \cos 2\pi n/ (M-1) + 0.08 \cos 4\pi n/(M-1), 0 \le n \le M-1$

= 0, otherwise

The window function of a non causal Blackman window is expressed by

$W_B(n) = 0.42 + 0.5 \cos 2\pi n/ (M-1) + 0.08 \cos 4\pi n/(M-1), 0 \le |n| \le (M-1)/2$

= 0, otherwise

The width of the main lobe is approximately 12π/M and the peak of the first side lobe is at -58dB.

11. What is the condition for linear phase of a digital filter? (APR 2005 ITDSP)

$h(n) = h(M-1-n) \rightarrow$ Linear phase FIR filter with a nonzero response at $\omega=0$

$h(n) = -h(M-1-n) \rightarrow$Low pass Linear phase FIR filter with a nonzero response at $\omega=0$

12. Define backward and forward predictions in FIR lattice filter. (NOV 2005 IT)

The reflection coefficient in the lattice predictor is the negative of the cross correlation coefficients between forward and backward prediction errors in the lattice.

13. List the important characteristics of physically realizable filters. (NOV 2005 ITDSP)

Symmetric and anti- symmetric

- Linear phase frequency response
- Impulse invariance

14. **Write the magnitude function of Butterworth filter. What is the effect of varying order of N on magnitude and phase response? (Nov/Dec2005) -ECE**

$|H(j\Omega)|^2 = 1 / [1 + (\Omega/\Omega_C)^{2N}]$ where N= 1,2,3,….

15. **List the characteristics of FIR filters designed using window functions. NOV 2004 ITDSP**

- the Fourier transform of the window function $W(e^{jw})$ should have a small width of main lobe containing as much of the total energy as possible
- the fourier transform of the window function $W(e^{jw})$ should have side lobes that decrease in energy rapidly as w to π. Some of the most frequently used window functions are described in the following sections

16. **Give the Kaiser Window function. (Apr/May 2004)-ECE**

The Kaiser Window function is given by

$W_K(n) = I_0(\beta) / I_0(\alpha)$, for $|n| \leq (M-1)/2$

Where α is an independent variable determined by Kaiser.

$B = \alpha[1 – (2n/M-1)^2]$

17. What is meant by FIR filter? And why is it stable? (APR 2004 ITDSP)

FIR filter → Finite Impulse Response. The desired frequency response of a FIR filter can be represented as

$$H_d(e^{j\omega})= \sum_{n=-\infty}^{\infty} h_d(n)e^{-j\omega n}$$

If h(n) is absolutely summable(i.e., Bounded Input Bounded Output Stable). So, it is in stable.

18. Mention two transformations to digitize an analog filter. (APR 2004 ITDSP)

(i) Impulse-Invariant transformation techniques

(ii) Bilinear transformation techniques

19. Draw the direct form realization of FIR system. (NOV 2004 ITDSP)

20.Give the equation specifying Barlett and hamming window. (NOV 2004 ITDSP)

The transfer function of Barlett window

$w_B(n) = 1-(2|n|)/(N-1), ((N-1)/2)\geq n\geq -((N-1)/2)$

The transfer function of Hamming window

$w_{hm}(n) = 0.54+0.46\cos((2\pi n)/(N-1), ((N-1)/2)\geq n\geq -((N-1)/2)$ α = 0.54

UNIT-V - FINITE WORD LENGTH EFFECTS

1. Compare fixed point and floating point arithmetic. Nov/Dec 2008 CSE&MAY 2006 IT

Fixed Point Arithmetic	Floating Point Arithmetic
• It covers only the dynamic range. • Compared to FPA, accuracy is poor • Compared to FPA it is low cost and easy to design • It is preferred for real time operation system • Errors occurs only for multiplication • Processing speed is high • Overflow is rare phenomenon	• It covers a large range of numbers • It attains its higher accuracy • Hardware implementation is costlier and difficult to design • It is not preferred for real time operations. • Truncation and rounding errors occur both for multiplication and addition • Processing speed is low • Overflow is a range phenomenon

2.What are the errors that arise due to truncation in floating point numbers
Nov/Dec 2008
CSE

1.Quantization error
2.Truncation error
Et=Nt-N

3.What are the effects of truncating an infinite flourier series into a finite series?
Nov/Dec 2008
CSE

4. Draw block diagram to convert a 500 m/s signal to 2500 m/s signal and state the problem due to this conversion April/May2008
CSE

5.List errors due to finite world length in filter design April/May2008
CSE

- Input quantization error
- Product quantization error
- Coefficient quantization error

5. What do you mean by limit cycle oscillations in digital filter? Nov/Dec 2007 CSE

In recursive system the nonlinearities due to the finite precision arithmetic operations often cause periodic oscillations to occur in the output ,even when the input sequence is zero or some non zero constant value .such oscillation in recursive system are called limit cycle oscillation

7.Define truncation error for sign magnitude representation and for 2's complement Representation April/May 2008 IT&APR 2005 IT

Truncation is a process of discarding all bits less significant than least significant bit that is retained For truncation in floating point system the effect is seen only in mantissa.if the mantissa is truncated to b bits ,then the error satisfies

$0 \geq \epsilon > -2.2^{-b}$ for x >0 and

$0 \leq \epsilon < -2.2^{-b}$ for x <0

8. What are the types of limit cycle oscillation? April/May 2008 IT

i.Zero input limit cycle oscillation

ii.overflow limit cycle oscillation

9. What is meant by overflow limit cycle oscillations? (May/Jun 2006)

In fixed point addition, overflow occurs due to excess of results bit, which are stored at the registers. Due to this overflow, oscillation will occur in the system. Thus oscillation is called as an overflow limit cycle oscillation.

10. How will you avoid Limit cycle oscillations due to overflow in addition(MAY 2006 IT DSP)

Condition to avoid the Limit cycle oscillations due to overflow in addition

$|a_1|+|a_2|<1$

a_1 and a_2 are the parameter for stable filter from stability triangle.

11.**What are the different quantization methods? (Nov/Dec 2006)-ECE**

- amplitude quantization
- vector quantization
- scalar quantization

12.List the advantages of floating point arithmetic. (Nov/Dec 2006)-ECE

- Large dynamic range
- Occurrence of overflow is very rare
- Higher accuracy

13.Give the expression for signal to quantization noise ratio and calculate the improvement with an increase of 2 bits to the existing bit.

(Nov/Dec 2006, Nov/Dec 2005)-ECE

$SNR_{A/D} = 16.81+6.02b-20\log_{10}(R_{FS}/\sigma_x)$ dB.

With b = 2 bits increase, the signal to noise ratio will increase by 6.02 X 2 = 12dB.

14. What is truncation error? (APR 2005 ITDSP)

Truncation is an approximation scheme wherein the rounded number or digits after the pre-defined decimal position are discarded.

15. What are decimators and interpolators? (APR 2005 ITDSP)

Decimation is a process of reducing the sampling rate by a factor D, i.e., down-sampling. Interpolation is a process of increasing the sampling rate by a factor I, i.e., up-sampling.

16.What is the effect of down sampling on the spectrum of a signal?

(APR 2005 ITDSP) & (APR 2005 ITDSP)

The signal (n) with spectrum $X(\omega)$ is to be down sampled by the factor D. The spectrum $X(\omega)$ is assumed to be non-zero in the frequency interval $0\leq|\omega|\leq\pi$.

17.Give the rounding errors for fixed and floating point arithmetic.

(APR 2004 ITDSP)

A number x represented by b bits which results in b_R after being Rounded off. The quantized error ε_R due to rounding is given by

$$\varepsilon_R=Q_R(x)-x$$

where $Q_R(x)$ = quantized number(rounding error)

The rounding error is independent of the types of fixed point arithmetic, since it involves the magnitude of the number. The rounding error is symmetric about zero and falls in the range.

$$-((2^{-bT}-2^{-b})/2)\leq \varepsilon_R \leq((2^{-bT}-2^{-b})/2)$$

ε_R may be +ve or –ve and depends on the value of x.

The error ε_R incurred due to rounding off floating point number is in the range

$$-2^E.2^{-bR/2})\leq \varepsilon_R \leq 2^E.2^{-bR/2}$$

18.Define the basic operations in multirate signal processing.

(APR 2004 ITDSP)

The basic operations in multirate signal processing are

(i)Decimation

(ii)Interpolation

Decimation is a process of reducing the sampling rate by a factor D, i.e., down-sampling. Interpolation is a process of increasing the sampling rate by a factor I, i.e., up-sampling.

19. Define sub band coding of speech. (APR 2004 ITDSP) & (NOV 2003 ECEDSP) & (NOV 2005 ECEDSP)

Sub band coding of speech is a method by which the speech signal is subdivided into several frequency bands and each band is digitally encode separately. In the case of speech signal processing, most of its energy is contained in the low frequencies and hence can be coded with more bits then high frequencies.

20.What is the effect of quantization on pole locations? (NOV 2004 ITDSP)

$$D(z) = \prod_{k=1}^{N} (1-p_k z^{-1})$$

▲pk is the error or perturbation resulting from the quantization of the filter coefficients

21.What is an anti-imaging filter? (NOV 2004 ITDSP)

The image signal is due to the aliasing effect. In caseof decimation by M, there will be M-1 additional images of the input spectrum. Thus, the input spectrum $X(\omega)$ is band limited to the low pass frequency response. An anti-aliasing filter eliminates the spectrum of $X(\omega)$ in the range $(л/D\leq \omega \leq\pi$.

The anti-aliasing filter is LPF whose frequency response $H_{LPF}(\omega)$ is given by

$$H_{LPF}(\omega) = 1,\ |\omega|\leq л/M$$
$$= 0, \text{ otherwise.}$$

D → Decimator

22.What is a decimator? If the input to the decimator is x(n)={1,2,-1,4,0,5,3,2}, What is the output? (NOV 2004 ITDSP)

Decimation is a process of reducing the sampling rate by a factor D, I.e., down-sampling.
$x(n)=\{1,2,-1,4,0,5,3,2\}$
$D=2$
Output $y(n) = \{1,-1,0,3\}$

23.What is dead band? (Nov/Dec 2004)-ECE

In a limit cycle the amplitude of the output are confined to a range of value, which is called dead band.

24.How can overflow limit cycles be eliminated? (Nov/Dec 2004)-ECE

- Saturation Arithmetic
- Scaling

25.What is meant by finite word length effects in digital filters? (Nov/Dec 2003)-ECE

The digital implementation of the filter has finite accuracy. When numbers are represented in digital form, errors are introduced due to their finite accuracy. These errors generate finite precision effects or finite word length effects.

When multiplication or addition is performed in digital filter, the result is to be represented by finite word length (bits). Therefore the result is quantized so that it can be represented by finite word register. This quantization error can create noise or oscillations in the output. These effects are called finite word length effects.

PART B

UNIT-1 - SIGNALS AND SYSTEMS

1.Determine whether the following signals are Linear ,Time Variant, causal and stable

(1) $Y(n)=\cos[x(n)]$ Nov/Dec 2008 CSE
(2) $Y(n)=x(-n+2)$
(3) $Y(n)=x(2n)$
(4) $Y(n)=x(n)+nx(n+1)$

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.79)

2. Determine the causal signal x(n) having the Z transform Nov/Dec 2008 CSE

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

Refer book : Digital signal processing by Ramesh Babu .(Pg no 2.66)

3. Use convolution to find x(n) if X(z) is given by Nov/Dec 2008 CSE

$$\frac{1}{\left(1-\frac{1}{2}z^{-1}\right)\left(1+\frac{1}{4}z^{-1}\right)}$$ **for ROC** $|z| > \frac{1}{2}$

Refer book : Digital signal processing by Ramesh Babu .(Pg no 2.62)

4.Find the response of the system if the input is {1,4,6,2} and impulse response of the system is {1,2,3,1}

April/May2008CSE

Refer book: Digital signal processing by A.Nagoor kani .(Pg no 23-24)

5.find r_{xy} and r_{yx} for x={1,0,2,3} and y={4,0,1,2}. April/May2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.79)

6. (i) Check whether the system y(n)=ay(n-1)+x(n) is linear ,casual, shift variant, and stable

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.51-1.57)

(ii) Find convolution of $\{5,4,\underset{\uparrow}{3},2\}$ and $\{1,0,3,\underset{\uparrow}{2}\}$ April/May2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.79)

7. (i) Compute the convolution y(n) of the signals

$$x(n)= \begin{cases} a^n, -3 \le n \le 5 \\ 0 \text{ , elsewhere} \end{cases}$$

and

$$h(n)= \begin{cases} 1, 0 \le n \le 4 \\ 0, \text{ elsewhere} \end{cases}$$

Nov/Dec 2007 CSE

8.A discrete-time system can be static or dynamic, linear or nonlinear, Time invariant or time varying, causal or non causal, stable or unstable. Examine the following system with respect to the properties also.

(1) y(n) = cos [x(n)]
(2) y(n)=x(-n+2)
(3) y(n)=x(2n)
(4) y(n)=x(n).cosWo(n)

Nov/Dec 2007 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.185-1.197)

9.(i) Determine the response of the casual system.
y(n)-y(n-1)=x(n)+x(n-1) to inputs x(n)=u(n) and x(n)=2-n u(n). Test its stability.
(ii) Determine the IZT of X(z)=1 / [(1-z-1)(1-z-1)2] Nov/Dec 2007 CSE

Refer book : Digital signal processing by A.Nagoor kani . (Pg no 463)

10.(i)Determine the Z-transform of the signal $x(n)=a^n u(n)-b^n u(-n-1)$, b>a and plot the ROC.

Refer John G Proakis and Dimtris G Manolakis, “Digital Signal Processing Principles, Algorithms and Application”, PHI/Pearson Education, 2000, 3rd Edition. Page number (157)

(ii) Find the steady state value given $Y(z)=\{0.5/[(1-0.75z^{-1})(1-z^{-1})]\}$

Refer John G Proakis and Dimtris G Manolakis, “Digital Signal Processing Principles, Algorithms and Application”, PHI/Pearson Education, 2000, 3rd Edition. Page number (207)

(iii) Find the system function of the system described by

$y(n)-0.75y(n-1)+0.125y(n-2)=x(n)-x(n-1)$ and plot the poles and zeroes of

11.(i) find the convolution and correlation for $x(n)=\{0,1,-2,3,-4\}$ and $h(n)=\{0.5,1,2,1,0.5\}$.

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.79)

(ii)Determine the Impulse response for the difference equation

$Y(n) + 3\,y(n-1)+2y(n-2)=2x(n)-x(n-1)$ April/May2008 IT

Refer book : Digital signal processing by Ramesh Babu .(Pg no 2.57)

12. (i) Compute the z-transform and hence determine ROC of x(n) where

$$X(n) = \begin{cases} (1/3)^{n} \quad u(n).n \geq 0 \\ (1/2)^{-n} \quad u(n).n<0 \end{cases}$$

Refer book : Digital signal processing by Ramesh Babu .(Pg no 2.20)

(iii) prove the property that convolution in Z-domains multiplication in time domain April/May2008 IT

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.77)

13.Find the response of the system if the input is {1,4,6,2} and impulse response of the system is {1,2,3,1} April/May2008CSE

Refer book: Digital signal processing by A.Nagoor kani .(Pg no 23-24)

14.find r_{xy} and r_{yx} for x={1,0,2,3} and y={4,0,1,2}. April/May2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.79)

15.(i) Check whether the system $y(n)=ay(n-1)+x(n)$ is linear ,casual, shift variant, and stable

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.51-1.57)

(ii) Find convolution of $\{5,4,\underset{\uparrow}{3},2\}$ and $\{1,0,3,\underset{\uparrow}{2}\}$ April/May2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.79)

16. (i) Compute the convolution y(n) of the signals

$$x(n)=\begin{cases} a^n, & -3\leq n\leq 5 \\ 0, & \text{elsewhere} \end{cases}$$

and

$$h(n)=\begin{cases} 1, & 0\leq n\leq 4 \\ 0, & \text{elsewhere} \end{cases}$$

Nov/Dec 2007 CSE

17.A discrete-time system can be static or dynamic, linear or nonlinear, Time invariant or time varying, causal or non causal, stable or unstable. Examine the following system with respect to the properties also.

(1) y(n) = cos [x(n)]
(2) y(n)=x(-n+2)
(3) y(n)=x(2n)
(4) y(n)=x(n).cosWo(n)

Nov/Dec 2007 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 1.185-1.197)

18.(i) Determine the response of the casual system.
y(n)-y(n-1)=x(n)+x(n-1) to inputs x(n)=u(n) and x(n)=2-n u(n). Test its stability.
(ii) Determine the IZT of X(z)=1 / [(1-z-1)(1-z-1)2] **Nov/Dec 2007 CSE**

Refer book : Digital signal processing by A.Nagoor kani .
(Pg no 463)

19.(i)Determine the Z-transform of the signal $x(n)=a^n u(n)-b^n u(-n-1)$, b>a and plot the ROC.

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (157)

(ii) Find the steady state value given $Y(z)=\{0.5/[(1-0.75z^{-1})(1-z^{-1})]\}$

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (207)

(iii) Find the system function of the system described by
y(n)-0.75y(n-1)+0.125y(n-2)=x(n)-x(n-1) and plot the poles and zeroes of H(z). **(MAY 2006 ITDSP)**

Refer signals and systems by P. Ramesh babu , page no:10.65
(To find the impulse response h(n) and take z-transform.)

20.(i)Using Z-transform, compute the response of the system y(n)=0.7y(n-1)-0.12y(n-2+x9n-1)+x(n-2) to the input x(n)=nu(n). Is the system stable?

Refer signals and systems by chitode, page no:4.99

(ii)State and prove the properties of convolution sum. **(MAY 2006 ECESS)**

Refer signals and systems by chitode, page no:4.43 to 4.45

21.State and prove the sampling theorem. Also explain how reconstruction of original signal is done from the sampled signal. (NOV 2006 ECESS)

Refer signals and systems by chitode, page no:3-2 to 3-7

22.Explain the properties of an LTI system. (NOV 2006 ECESS)

Refer signals and systems by chitode, page no:4.47 to 4.49

23.a. Find the convolution sum for the $x(n) =(1/3)^{-n}$ u(-n-1) and h(n)=u(n-1)

Refer signals and systems by P. Ramesh babu , page no:3.76,3.77

b. Convolve the following two sequences linearly x(n) and h(n) to get y(n).

x(n)= {1,1,1,1} and h(n) ={2,2}.Also give the illustration

Refer signals and systems by chitode, page no:67

c. Explain the properties of convolution. (NOV2006 ECESS)

Refer signals and systems by chitode, page no:4.43 to 4.45

24. Check whether the following systems are linear or not

1. **$y(n) = x^2(n)$**
2. **y(n) = nx(n) (APRIL 2005 ITDSP)**

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (67)

25.(i)Determine the response of the system described by,

y(n)-3y(n-1)-4y(n- 2)=x(n)+2x(n-1) when the input sequence is $x(n)=4^n$ u(n).

Refer signals and systems by P. Ramesh babu , page no:3.23

(ii)Write the importance of ROC in Z transform and state the relationship between Z transforms to Fourier transform. (APRIL 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (153)

Refer S Poornachandra & B Sasikala, "Digital Signal Processing", Page number (6.10)

UNIT-II - FAST FOURIER TRANSFORMS

1.By means of DFT and IDFT ,Determine the sequence x3(n) corresponding to the circular convolution of the sequence x1(n)={2,1,2,1}.x2(n)={1,2,3,4}. Nov/Dec 2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 3.46)

2. State the difference between overlap save method and overlap Add method Nov/Dec 2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 3.88)

3. Derive the key equation of radix 2 DIF FFT algorithm and draw the relevant flow graph taking the computation of an 8 point DFT for your illustration Nov/Dec 2008 CSE

Refer book : Digital signal processing by Nagoor Kani .(Pg no 215)

4. **Compute the FFT of the sequence x(n)=n+1 where N=8 using the in place radix 2 decimation in frequency algorithm. Nov/Dec 2008 CSE**

Refer book : Digital signal processing by Nagoor Kani .(Pg no 226)

5. Find DFT for {1,1,2,0,1,2,0,1} using FFT DIT butterfly algorithm and

plot the spectrum April/May2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 4.17)

6. (i)Find IDFT for {1,4,3,1} using FFT-DIF method April/May2008 CSE

(ii)Find DFT for {1,2,3,4,1} (MAY 2006 ITDSP)

Refer book : Digital signal processing by Ramesh Babu .(Pg no 4.29)

7.Compute the eight point DFT of the sequence x(n)={ ½,½,½,½,0,0,0,0} using radix2 decimation in time and radix2 decimation in frequency algorithm. Follow exactly the corresponding signal flow graph and keep track of all the intermediate quantities by putting them on the diagram. Nov/Dec 2007 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 4.30)

8.(i) Discuss the properties of DFT.

Refer book : Digital signal processing by S.Poornachandra.,B.sasikala.
(Pg no 749)

(ii)Discuss the use of FFT algorithm in linear filtering. Nov/Dec 2007 CSE

Refer book : Digital signal processing by John G.Proakis .(Pg no 447)

9.(i) if x(n) $\xleftrightarrow{\text{N pt DFT}}$ **X(k) then, prove that**

X1(n)x2(n)=1/N [Xt(k) X2 Ⓝ**]. April/May2008 IT**

Refer book : Digital signal processing by Ramesh Babu .(Pg no 3.34)

(ii) Find 8 Point DFT of x(n)=0.5,0≤n≤3 Using DIT FFT
0, 4≤n≤7 April/May2008 IT

Refer book : Digital signal processing by Ramesh Babu .(Pg no 4.32)

10.Derive the equation for radix 4 FFT for N=4 and Draw the butterfly Diagram.
April/May2008 IT

11. (i) Compute the 8 pt DFT of the sequence
x(n)={0.5,0.5,0.5,0.5,0,0,0,0} using radix-2 DIT FFT

Refer P. Ramesh babu, "Signals and Systems".Page number (8.89)

(ii) Determine the number of complex multiplication and additions involved in a N-point Radix-2 and Radix-4 FFT algorithm. (MAY 2006 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (456 & 465)

12.Find the 8-pt DFT of the sequence x(n)={1,1,0,0} (APRIL 2005 ITDSP)

Refer P. Ramesh babu, "Signals and Systems". Page number (8.58)

13.Find the 8-pt DFT of the sequence
x(n)= 1, 0≤n≤7
0, otherwise
using Decimation-in-time FFT algorithm (APRIL 2005 ITDSP)

Refer P. Ramesh babu, "Signals and Systems".Page number (8.87)

14.Compute the 8 pt DFT of the sequence
x(n)={0.5,0.5,0.5,0.5,0,0,0,0} using DIT FFT (NOV 2005 ITDSP)

Refer P. Ramesh babu, "Signals and Systems".Page number (8.89)

15.By means of DFT and IDFT , determine the response of an FIR filter with impulse response h(n)={1,2,3},n=0,1,2 to the input sequence x(n) ={1,2,2,1}.
(NOV 2005 ITDSP)

Refer P. Ramesh babu, “Signals and Systems”.Page number (8.87)

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

Refer P. Ramesh babu, “Signals and Systems”.Page number (8.58)

(ii)Find the output sequence y(n) if h(n)={1,1,1} and x(n)={1,2,3,4} using circular convolution (APR 2004 ITDSP)

Refer P. Ramesh babu, “Signals and Systems”.Page number (8.65)

17. (i)What is decimation in frequency algorithm? Write the similarities and differences between DIT and DIF algorithms. (APR 2004 ITDSP) & (MAY 2006 ECEDSP)

Refer P. Ramesh babu, “Signals and Systems”. Page number (8.70-8.80)

18.Determine 8 pt DFT of x (n)=1for -3≤n≤3 using DIT-FFT algorithm (APR 2004 ITDSP)

Refer P. Ramesh babu, “Signals and Systems”. Page number (8.58)

19.Let X(k) denote the N-point DFT of an N-point sequence x(n).If the DFT of X(k)is computed to obtain a sequence $x_1(n)$. Determine $x_1(n)$ in terms of x(n) (NOV 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, “Digital Signal Processing Principles, Algorithms and Application”, PHI/Pearson Education, 2000, 3rd Edition. Page number (456 & 465)

UNIT-III - IIR FILTER DESIGN

1.Design a digital filter corresponding to an analog filter H(s)=$\frac{0.5(s+4)}{(s+1)(s+4)}$ using the impulse invariant method to work at a sampling frequency of 100 samples/sec
Nov/Dec 2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no5.40)

2.Determine the direct form I ,direct form II ,Cascade and parallel structure for the system
Y(n)=-0.1y(n-1)+0.72y(n-2)+0.7x(n)-0.25x(n-2) Nov/Dec 2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no5.61)

3.What is the main drawback of impulse invariant method ?how is this overcome by bilinear transformation? Nov/Dec 2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no5.46)

4.Design a digital butter worth filter satisfying the constraints Nov/Dec 2008 CSE

0.707≤$|H(e^{jw}|$≤1 for 0 ≤w≤$\frac{\pi}{2}$

$|H(e^{jw}|$≤0.20 for $3\frac{\pi}{4}$ ≤w≤ π

With T=1 sec using bilinear transformation .realize the same in Direct form II

Refer book : Digital signal processing by Ramesh Babu .(Pg no5.79)

5. (i)Design digital filter with H(s) =$1/(s2+7s+12)$ using T=1sec.

(ii)Design a digital filter using bilinear transform for H(s)=2/(s+1)(s+2)with cutoff frequency as 100 rad/sec and sampling time =1.2 ms April/May2008 CSE

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 341)

6. (i) Realize the following filter using cascade and parallel form with

direct form –I structure

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

(ii) Find H(s) for a 3rd order low pass butter worth filter April/May2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 5.8)

7.(i) Derive bilinear transformation for an analog filter with system function
H(s) =b / (s+a)

Refer book: Digital signal processing by John G.Proakis .(Pg no 676-679)

(ii)Design a single pole low pass digital IIR filter with -3 db bandwidth of 0.2п by use of bilinear transformation. Nov/Dec 2007 CSE

8.(i) Obtain the Direct Form I, Direct Form II, cascade and parallel realization for the following system Y(n)= -0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)

Refer book : Digital signal processing by Ramesh Babu .(Pg no 5.68)

(ii) Discuss the limitation of designing an IIR filter using impulse invariant method. Nov/Dec 2007 CSE

Refer book : Digital signal processing by A.Nagoor kani . (Pg no 330)

9. Design a low pass Butterworth filter that has a 3 dB cut off frequency of 1.5 KHz and an attenuation of 40 dB at 3.0 kHz April/May2008 IT

Refer book : Digital signal processing by Ramesh Babu .(Pg no 5.14)

10. (i) Use the Impulse invariance method to design a digital filter from an analog prototype that has a system function

April/May2008 IT

$Ha(s)=s+a/((s+a)^2+b^2)$

Refer book : Digital signal processing by Ramesh Babu .(Pg no 5.42)

(ii) Determine the order of Cheybshev filter that meets the following specifications

(1) 1 dB ripple in the pass band 0≤|w| ≤ 0.3 b
(2) Atleast 60 dB attrnuation in the stop band 0.35∏ ≤|w| ≤∏ Use Bilinear Transformation

Refer book : Digital signal processing by Ramesh Babu .(Pg no 5.27)

11.(i) Convert the analog filter system function$H_a(s)=\{(s+0.1)/[(s+0.1)^2+9]\}$ into a digital IIR filter using impulse invariance method.(Assume T=0.1sec) (APR 2006 ECEDSP)

Refer John G Proakis and Dimtris G Manolakis, “Digital Signal Processing Principles, Algorithms and Application”, PHI/Pearson Education, 2000, 3rd Edition. Page number (675)

12.Determine the Direct form II realization for the following system:
y(n)=-0.1y(n-1+0.72y(n-2)+0.7x(n)-0.252x(n-2). (APRIL 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, “Digital Signal Processing Principles, Algorithms and Application”, PHI/Pearson Education, 2000, 3rd Edition. Page number (601-7.9b)

13.Explain the method of design of IIR filters using bilinear transform method.

(APRIL 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (676-8.3.3)

14.Explain the following terms briefly:

(i)Frequency sampling structures

(ii)Lattice structure for IIR filter (NOV 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (506 &531)

15.Consider the system described by

y(n)-0.75y(n-1)+0.125y(n-2)=x(n)+0.33x(n-1).

Determine its system function (NOV 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (601-7.37)

16.Find the output of an LTI system if the input is $x(n)=(n+2)$ for $0\leq n\leq 3$ and $h(n)=a^n u(n)$ for all n (APR 2004 ITDSP)

Refer signals and systems by P. Ramesh babu , page no:3.38

17.Obtain cascade form structure of the following system:

y(x)=-0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2) (APR 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (601-7.9c)

18.Verify the Stability and causality of a system with

$H(z)=(3-4Z^{-1})/(1+3.5Z^{-1}+1.5Z^{-2})$ (APR 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (209)

UNIT-IV - FIR FILTER DESIGN

1.Design a FIR linear phase digital filter approximating the ideal frequency response

Nov/Dec 2008 CSE

$$Hd(w) = \begin{Bmatrix} 1, for\ |w| \leq \frac{\pi}{6} \\ 0, for\ \frac{\pi}{6} < |w| \leq \pi \end{Bmatrix}$$

With T=1 Sec using bilinear transformation .Realize the same in Direct form II

Refer book : Digital signal processing by Nagoor Kani .(Pg no 367)

2.Obtain direct form and cascade form realizations for the transfer function of the system given by

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

Nov/Dec 2008 CSE

Refer book : Digital signal processing by Nagoor Kani .(Pg no 78)

3.Explain the type I frequency sampling method of designing an FIR filter.

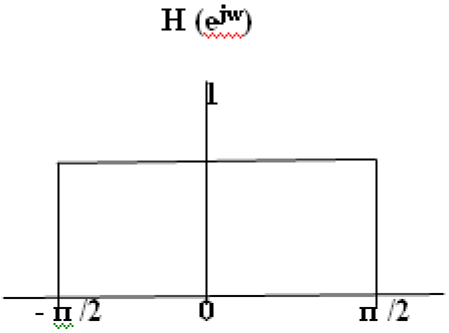
Nov/Dec 2008
CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no6.82)

4.Compare the frequency domain characteristics of various window functions .Explain how a linear phase FIR filter can be used using window method. Nov/Dec 2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no6.28)

5. Design a LPF for the following response .using hamming window with N=7



April/May2008 CSE

6. (i) Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n)= \pm h(M-1-n)$, n=0,1,….M-1. Also discuss symmetric and antisymmetric cases of FIR filter. Nov/Dec 2007

Refer book: Digital signal processing by John G.Proakis .
(Pg no 630-632)

(ii) Explain the need for the use of window sequences in the design of FIR filter. Describe the window sequences generally used and compare their properties.

Nov/Dec 2007 CSE

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 292-295)

7.(I) Explain the type 1 design of FIR filter using frequency sampling technique. Nov/Dec 2007 CSE

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 630-632)

(ii)A low pass filter has the desired response as given below

$$Hd(e^{jw})= \begin{cases} e^{-i3w}, & 0 \le w < \prod/2 \\ 0, & \prod/2 \le < \prod \end{cases}$$

Determine the filter coefficients h(n) for M=7 using frequency sampling method.

Nov/Dec 2007
CSE

8.(i) For FIR linear phase Digital filter approximating the ideal frequency response

$$Hd(w) = 1 \quad \le |w| \le \prod /6$$

$$0 \quad \prod /6 \le |w| \le \prod$$

Determine the coefficients of a 5 tap filter using rectangular Window

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 415

(ii) Determine the unit sample response h(n) of a linear phase FIR filter of Length M=4 for which the frequency response at w=0 and w= ∏/2 is given as Hr(0) ,Hr(∏/2) =1/2

April/May2008 IT

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 310)

9.(i) Determine the coefficient h(n) of a linear phase FIR filter of length M=5 which has symmetric unit sample response and frequency response

Hr(k)=1 for k=0,1,2,3
0.4 for k=4
0 for k=5, 6, 7 April/May2008 IT(NOV 2005 ITDSP)

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 308)

(ii) Show that the equation $\sum_{n=0}^{m-1}$ h(n)=sin (wj-wn)=0,is satisfied for a linear phase FIR filter of length 9

April/May2008 IT

10. Design linear HPF using Hanning Window with N=9

H(w) =1 -п to Wc and Wc to п
= 0 otherwise

April/May2008 IT

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 301)

11.Explain in detail about frequency sampling method of designing an FIR filter.
(NOV 2004 ITDSP) & (NOV 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (630)

12.Explain the steps involved in the design of FIR Linear phase filter using window method.
(APR 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (8.2.2 & 8.2.3)

13.(i)What are the issues in designing FIR filter using window method?

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (8.2)

(ii)An FIR filter is given by
y(n)=2x(n)+(4/5)x(n-1)+(3/2)x(n-2)+(2/3)x(n-3) find the lattice structure coefficients (APR 2004 ITDSP)

Refer S Poornachandra & B Sasikala, "Digital Signal Processing", Page number (FIR-118)

UNIT-V - FINITE WORD LENGTH EFFECTS

1.Draw the circuit diagram of sample and hold circuit and explain its operation
Nov/Dec 2008 CSE/ Nov/Dec 2007 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no1.172)

2. The input of the system y(n)=0.99y(n-1)+x(n) is applied to an ADC .what is the power produced by the quantization noise at the output of the filter if the input is quantized to 8 bits

Nov/Dec 2008 CSE

Refer book : Digital signal processing by Nagoor Kani .(Pg no 423)

3.Discuss the limit cycle in Digital filters Nov/Dec 2008 CSE

Refer book : Digital signal processing by Nagoor Kani .(Pg no 420)

4.What is vocoder? Explain with a block diagram Nov/Dec 2008 CSE/ Nov/Dec 2007 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no10.7)

(ii) Discuss about multirate Signal processing April/May 2008 CSE

Refer book : Digital signal processing by Ramesh Babu .(Pg no 8.1)

5. (i) Explain how the speech compression is achieved .

(ii) Discuss about quantization noise and derive the equation for finding quantization noise power. April/May2008CSE

Refer book : Digital signal processing by Ramesh Babu.(Pg no 7.9-7.14)

6. Two first order low pass filter whose system functions are given below are connected in cascade. Determine the overall output noise power. H1(z) = 1/ (1-0.9z-1) and H2(z) = 1/ (1-0.8z-1) Nov/Dec 2007 CSE

Refer book: Digital signal processing by Ramesh Babu. (Pg no 7.24)

7. Describe the quantization errors that occur in rounding and truncation in two's complement. Nov/Dec 2007 CSE

Refer book : Digital signal processing by John G.Proakis .(Pg no 564)

8. Explain product quantization and prove $\sigma_{err}^2 = \sum_{i=1}^{m} \sigma^2_{oi}$ April/May2008 IT

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 412)

9.A cascade Realization of the first order digital filter is shown below ,the system function of the individual section are $H1(z)=1/(1-0.9z^{-1})$ and $H2(z)=1/(1-0.8z^{-1})$ .Draw the product quantization noise model of the system and determine the overall output noise power

April/May2008 IT

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 415)

9. (i) Show dead band effect on y(n) = .95 y(n-1)+x(n) system restricted to 4 bits .Assume x(0) =0.75 and y(-1)=0

Refer book : Digital signal processing by A.Nagoor kani .(Pg no 423-426)

11. Explain the following terms briefly:

(i)Perturbation error

(ii)Limit cycles (NOV 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number(7.7.1 &7.7.2)

12.(i) Explain clearly the downsampling and up sampling in multirate signal processing. (APRIL 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing

Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (784-790)

(ii)Explain subband coding of speech signal

(NOV 2003 ITDSP) & (NOV 2004 ITDSP) & (NOV 2005 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number(831-833)

13.(i) Derive the spectrum of the output signal for a decimator

(ii) Find and sketch a two fold expanded signal y(n) for the input

(APR 2004 ITDSP) &(NOV 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (788)

14.(i)Propose a scheme for sampling rate conversion by a rational factor I/D.

(NOV 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (790)

15. Write applications of multirate signal processing in Musical sound processing

(NOV 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (952)

16. With examples illustrate (i) Fixed point addition (ii) Floating point multiplication (iii) Truncation (iv) Rounding.(APR 2005 ITDSP) & (NOV 2003 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (7.5)

17. Describe a single echo filter using in musical sound processing.

(APRIL 2004 ITDSP)

Refer John G Proakis and Dimtris G Manolakis, "Digital Signal Processing Principles, Algorithms and Application", PHI/Pearson Education, 2000, 3rd Edition. Page number (12.5.3)

STAFF INCHARGE **HOD/ECE**