JEPPIAAR ENGINEERING COLLEGE

QUESTION BANK

CS 2403 – DIGITAL SIGNAL PROCESSING

UNIT I SIGNALS AND SYSTEMS 9

Basic elements of DSP – concepts of frequency in Analog and Digital Signals – sampling theorem – Discrete – time signals, systems – Analysis of discrete time LTI systems – Z transform – Convolution (linear and circular) – Correlation.

UNIT II FREQUENCY TRANSFORMATIONS 9

Introduction to DFT – Properties of DFT – Filtering methods based on DFT – FFT Algorithms Decimation – in – time Algorithms, Decimation – in – frequency Algorithms – Use of FFT in Linear Filtering – DCT.

UNIT III IIR FILTER DESIGN 9

Structures of IIR – Analog filter design – Discrete time IIR filter from analog filter – IIR filter design by Impulse Invariance, Bilinear transformation, Approximation of derivatives – (HPF, BPF, BRF) filter design using frequency translation

UNIT IV FIR FILTER DESIGN 9

Structures of FIR – Linear phase FIR filter – Filter design using windowing techniques, Frequency sampling techniques – Finite word length effects in digital Filters

UNIT V APPLICATIONS 9

Multirate signal processing – Speech compression – Adaptive filter – Musical sound processing – Image enhancement.

TEXT BOOKS:

1. John G. Proakis & Dimitris G.Manolakis, “Digital Signal Processing – Principles, Algorithms & Applications”, Fourth edition, Pearson education / Prentice Hall, 2007.
2. Emmanuel C..Ifeachor, & Barrie.W.Jervis, “Digital Signal Processing”, Second edition, Pearson Education / Prentice Hall, 2002.

REFERENCES:

3. Alan V.Oppenheim, Ronald W. Schafer & Hohn. R.Back, “Discrete Time Signal Processing”, Pearson Education.

4. Andreas Antoniou, “Digital Signal Processing”, Tata McGraw Hill.

UNIT I - SIGNALS ANS SYTEMS

1. **What are energy and power signals?**
   The energy signal is one in which has finite energy and zero average power. The power signal is one in which has finite average power and infinite energy.

$$E = \lim_{T\to\infty} \int_{-T}^{T} |x(t)|^2 \, dt \text{ joules}$$

$$P = \lim_{T\to\infty} 1/2T \int_{-T}^{T} |x(t)|^2 \, dt \text{ joules}$$

2. **What are Random signals?**
   The signal that takes on random value at any given instant and it cannot be predicted. Non deterministic signals are also called as Random signals.
   Example: ECG signals, Noise in electric circuits.

3. **Distinguish between continuous and discrete time signals.**
   If the signal amplitude can be defined for all values of time t is Called as continuous time signals and it is denoted by x(t).
   If the signal amplitude can be defined for particular integer values of time period n is called as discrete time signals and it is denoted by x(n).

4. **Find the period of x(n) = cos [8πn/7 +2].**
   $\omega = 8\pi/7$
   $2\pi f = 8\pi/7$
   $f = 4/7$; here K= 4 & N =7
   It is periodic and the fundamental period is N =7 samples.

5. **Define System.**
   A system is a physical device that performs an operation on the signal. The input signal is called as excitation and output signal is called as response.

6. **State the classification of system.** **(Nov/Dec 2004)**
   I. Linear and non linear system

II. Time variant and in variant system
III. Causal and non causal system.
IV. Static and Dynamic system.
V. Stable and unstable system.

7. Distinguish between linear and non linear system.

$$a_1 y_1(t) + a_2 y_2(t) = f[a_1x_1(t) + a_2x_2(t)]$$

If the above equation satisfies then the system is said to be linear system. If the above equation does not satisfy then the system is said to be non linear system.

8. State the properties of linear system.

A system is said to be linear it should satisfy the principle of Superposition.

9. Define time invariant system. **(May/June 2006)**

A system is time invariant if it satisfies the following relation $F[x(t - t_1)] = y(t - t_1)$.

10. What is meant by causal & non causal system? **(May/June 2006)**

A system is said be causal if it's output at anytime depends upon present and past input only. A system is said be non causal if it's output at anytime depends upon present and future input only.

11. State the condition for the BIBO stable? **May/June 2005**

The condition for the BIBO stable is given by

$$\int^{\infty} |h(t)|\, dt < ^{a}$$

12. Define Z- Transform.

The Z – transform of a discrete time signal x(n) is defined as the power series

$$Z[x(n)] = X(z) = \sum_{n=-\alpha}^{\alpha} x(n)\, z^{-n}$$

13. Define Sampling Theorem. **Nov/Dec 2008**

Sampling is the process by which analog signal is converted into corresponding sequence of samples that are spaced uniformly in time.

14.What is aliasing?

The phenomenon of high frequency sinusoidal components acquiring the identity of low frequency sinusoidal components after sampling is called aliasing.

15. **Check whether the system $y(n) = e^{x(n)}$ is linear.** **May/June 2007**

Consider two signals $x_1(n)$ and $x_2(n)$.Let $y_1(n)$ & $y_2(n)$ be the response of the system for inputs $x_1(n)$ and $x_2(n)$ respectively.

$y_1(n) = H[x_1(n)] = e^{x_1(n)}$

$y_2(n) = H[x_2(n)] = e^{x_2(n)}$

$y_3(n) = H[a_1x_1(n) + a_2 x_2(n)] = e^{(a_1x_1(n) + a_2x_2(n))} = e^{a_1x_1(n)} . e^{a_2x_2(n))}$

Therefore, $a_1y_1(n) + a_2y_2(n) = a_1e^{x_1(n)} + a_2\, e^{x_2(n)}$

$y_3(n)$ is not equal to $a_1y_1(n) + a_2y_2(n)$. Hence the system is non linear.

PART-B

1) **Determine whether the following signals are linear,time variant, causal and stable.** **Nov/Dec 2008**

$Y(n) = Cos[x(n)]$

$Y(n) = x(-n+2)$

$Y(n) = x(2n)$

$Y(n) = x(n) + nx(n+1)$

Ans: Ref Pg.No 31-45 , DSP by Nagoor Kani.A

2) **Find the response of the system for the input signal $X(n) = \{1,2,2,3\}$ and $h(n) = \{1,0,3,2\}$** **May/June 2007**

Ans: Ref Pg.No 164 , DSP by Nagoor Kani.A

3) **Explain sampling theorem**

Ans: Ref Pg.No 26 , DSP by Proakis

4) **Find the inverse z- transform of** **May/June 2007**

$1/ (1-0.5 z^{-1}) (1-z^{-1})$

Ans: Ref Pg.No 461, DSP by Nagoor Kani.A

5) **Perform circular convolution of the two sequences.**

$X_1(n) = \{2,1,2,1\}$

$X_2(n) = \{1,2,3,4\}$

Ans: Ref Pg.No 175, DSP by Nagoor Kani.A

UNIT II - FREQUENCY TRANSFORMATION

PART-A

1. **State and prove Parseval's Theorem.** **Nov/Dec 2007**

Parseval's theorem states that

If $x(n) \leftrightarrow X(K)$ and $y(n) \leftrightarrow Y(K)$, Then

$$\sum_{n=0}^{N-1} x(n)\, y^*(n) = 1/N \sum_{K=0}^{N-1} X(K)\, Y^*(K)$$

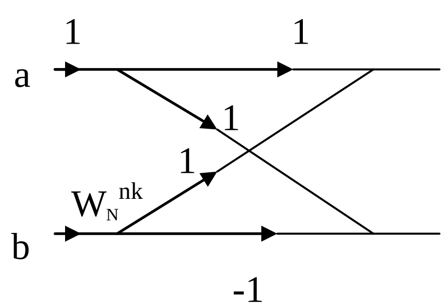
When y(n) = x(n), the above equation becomes

$$\boxed{\sum_{n=0}^{N-1} |x(n)|^2 = 1/N \sum_{k=0}^{N-1} |X(K)|^2}$$

2. **. What do you mean by the term "bit reversal" as applied to FFT?** **Nov/Dec 2007**

Re-ordering of input sequence is required in decimation – in –time. When represented in binary notation sequence index appears as reversed bit order of row number.

3. **Draw the basic butterfly diagram of radix -2 FFT.** **April/May 2008.**



$A = a + W_N^{nk} b$

$B = a - W_N^{nk} b$

4. **Distinguish between DIT and DIF –FFT algorithm.** **Nov/Dec 2008**

S.No	DIT –FFT Algorithm	DIF –FFT Algorithm
1.	The input is in bit reversed order; the output will be normal order.	The input is in normal order; the output will be bit reversed order.
2.	Each stage of computation the phase factor are multiplied before add subtract operation.	Each stage of computation the phase factor are multiplied after add subtract operation.

5. **If H(K) is the N-point DFT of a sequence h(n) , Prove that H(K) and H(N-K) are comples conjugates.** **Nov/Dec 2008**

This property states that, if h(n) is real , then

$H(N-K) = H^*(K) = H(-K)$

Proof:

By the definition of DFT;

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

Replace 'K' by 'N-K'

$$X(N-K) = \sum^{N-1} x(n)\, e^{(-j2\pi n(N-K))/N}$$

$$\boxed{X(N-K) = X^*(K)}$$

6. **The first five DFT coefficients of a sequence x(n) are X(0) = 20, X(1) = 5+j2,X(2) = 0,X(3) = 0.2+j0.4 , X(4) = 0 . Determine the remaining DFT coefficients.** **May/June 2007**

Solution:

$X(K) = [20, 5+j2, 0, 0.2+j0.4, 0, X(5), X(6), X(7)]$

$X(5) = 0.2 - j0.4$

$X(6) = 0$

$X(7) = 5-j2$

7. **What is FFT?** **Nov/Dec 2006**

The Fast Fourier Transform is a method or algorithm for computing the DFT with reduced number of calculations. The computational efficiency can be achieved if we adopt a divider and conquer approach. This approach is based on decomposition of an N-point DFT in to sucessively smaller DFT's. This approach leads to a family of an efficient computational algorithm is known as **FFT algorithm**.

8. **What are the advantages of FFT algorithm over direct computation of DFT?** **May/June 2007**

   1. Reduces the computation time required by DFT.
   2. Complex multiplication required for direct computation is N^2 and for FFT calculation is $N/2 \log_2 N$.

Speed calculation

9. **Define the properties of convolution.** **April/May 2008.**
    3. **Commutative property:** $x(n)*h(n) = h(n)*x(n)$
    4. **Associative Property:** $[x(n)*h_1(n)]*h_2(n) = x(n)*[h_1(n)*h_2(n)]$
    5. **Distributive Property:** $x(n)*[h_1(n)+h_2(n)] = [x(n)*h_1(n)]+[x(n)*h_1(n)]$

10. **Distinguish between linear & circular convolution.**

s.no	Linear convolution	circular convolution
1	The length of the input sequence can be different.	The length of the input sequence should be same.
2	Zero Padding is not required.	Zero padding is required if the length of the sequence is different.

11. **Define Zero padding? Why it is needed?**
Appending zeros to the sequence in order to increase the size or length of the sequence is called zero padding.In circular convolution , when the two input sequence are of different size , then they are converted to equal size by zero padding.

12. **State the shifting property of DFT.**
Time shifting property states that

$$\text{DFT}\{x(n-n_0)\} = X(K)\, e^{(-j2\pi n0k)/N}$$

13. **Why do we go for FFT?**
The FFT is needed to compute DFT with reduced number of calculations.The DFT is required for spectrum analysis on the sinals using digital computers.

14. **What do you mean by radix-2 FFT?**
The radix -2 FFT is an efficient algorithm for coputing N- point DFT of an N-point sequence .In radix-2 FFT the n-point is decimated into 2-point sequence and the 2-point DFT for each decimated sequence is computed. From the results of 2-point DFT's, the 4-point DFT's are computed. From the results of 4 –point DFT's ,the 8-point DFT's are computed and so on until we get N - point DFT.

15. **Give any two application of DFT?**
    1. The DFT is used for spectral analysis of signals using a digital computer.
    2. The DFT is used to perform filtering operations on signals using digital computer.

16. **How many multiplications & addition are involved in radix-2 FFT?**
    For performing radix-2 FFT, the value of Nshould be such that, $N = 2^m$. The total numbers of complex additions are $N\log_2 N$ and the total number of complex multiplication are $(N/2)\log_2 N$.

17. **What is Twiddle factor?**
    Twiddle factor is defined as $W_N = e^{-j2\pi/N.}$ It is also called as weight factor.

18. **What is main advantage of FFT?**
    FFT reduces the computation time required to compute Discrete Fourier Transform.

PART-B

1. Derive the equation for Decimation – in time algorithm for FFT. Nov/Dec 2006 & April /May 2008 & Nov/Dec 2008

Ans: Ref Pg.No 207-209 , DSP by Nagoor Kani.A

2. (i)From first principles obtain the signal flow graph for Computing 8-point using radix -2 DIF –FFT algorithm.
ii) Using the above signal flow graph compute DFT of $x(n) = \cos(n\pi/4)$,$0 \le n \le 7$.
May/June 2007 & Nov/Dec 2007 & Nov/Dec 2008

Ans: Ref Pg.No 219 , DSP by Nagoor Kani.A
X(K) = {0, 3, 0, 2.7-j0.7, 0, 1, 0, 1.293-j0.7}

3. i)Discuss in detail the important properties of the DFT.
ii)Find the 4-point DFT of the sequence $x(n) = \cos(n\pi/4)$
iii)Compute an 8-point DFT using DIF FFT radix -2 algorithm.
x(n) = { 1,2,3,4,4,3,2,1} April /May 2008

Ans: i)Ref Pg.No 308-311, DSP by Salivahanan.
ii) X(K) = {1, 1-j1.414, 1, 1+j1.414}
X(K) = {20,-5.8-j2.4, 0, 0.17-j0.414, 0, -0.17+j0.414, 0,-5.82+j2.414}.

4. i)Prove the following properties of DFT when H(k) is the DFT of an N-point sequence h(n). H(k) is real and even when h(n) is real and even.H (k) is imaginary and odd when h(n) is real and odd.Compute the DFT of $x(n) = e^{-0.5n}$, $0 \le n \le 5$.

May/June 2007

Ans: i) Ref Pg.No 309, DSP by Salivahanan.
X(K) = { 2.414, 0.87-j0.659, 0.627-0.394j, 1.202,0.62-j0.252, 0.627-j0.252}.

5. Two finite duration sequence are given by
$x(n) = \sin (n\pi/2)$ for n = 0,1,2,3
$h(n) = 2^n$ for n = 0,1,2,3 Determine circular convolution using DFT &IDFT method. **Nov/Dec 2007**

Ans: X(K) = {0, -2j, 0, 2j}
H(K) = {15, -3+6j, -5, -3-6j}
y(n) = {6, -3, -6, 3}

6. Calculate the DFT of the sequence x(n) = {1,1,-2,-2}.Determine the response of LTI system by radix -2 DIT FFT. **Nov/Dec 2006**

Ans:i) X(K) = { 0, -1-j,6,-1+j}
Ref Pg.No 320-328, DSP by Salivahanan

UNIT III - IIR FILTER DESIGN

1. What are the different types of filters based on impulse response?

Based on impulse response the filters are of two types
1. IIR filter
2. FIR filter

The IIR filters are of recursive type, whereby the present output sample depends on the present input, past input samples and output samples. The FIR filters are of non recursive type, whereby the present output sample depends on the present input sample and previous input samples.

2. What are the different types of filters based on frequency response?

Based on frequency response the filters can be classified as
1. Lowpass filter
2. Highpass filter
3. Bandpass filter
4. Bandreject filter

3. **Distinguish between FIR filters and IIR filters.**

FIR filter IIR filter

1. These filters can be easily designed tohave perfectly linear phase.These filters do not have linear phase.
2. FIR filters can be realized recursive and non-recursively.
3. Greater flexibility to control the shapeof their magnitude response.
4. Errors due to round off noise are less severe in FIR filters, mainly because feedback is not used.IIR filters are easily realized recursively.Less flexibility, usually limited to specific
kind of filters.The round off noise in IIR filters is
more.

4. **What are the design techniques of designing FIR filters?**

There are three well known methods for designing FIR filters with linear phase .They are (1.)Window method (2.) Frequency sampling method (3.) Optimal or minimax design.

5. **What is Gibb's phenomenon?**

One possible way of finding an FIR filter that approximates H(ejw) would be to truncate the infinite Fourier series at n=±(N-1/2).Direct truncation of the series will lead to fixed percentage overshoots and undershoots before and after an approximated discontinuity in the frequency response.

6. **Find the digital transfer function H(Z) by using impulse invariant method for the analog transfer function H(S) = 1/ (S+2).Assume T=0.5sec** **May /June 2007 &Nov/Dec 2007**

Solution:

$H(S) = 1/ (S+2).$

$H(Z) = 1/[1-e^{-1} Z^{-1}]$

$H(Z) = 1/ [1-0.368Z^{-1}]$

7. **What is the relationship between analog and digital frequency in impulse invariant transformation?** **April/May 2008**

Digital Frequency: $\omega = \Omega T$

Ω = analog frequency

T= Sampling interval

8. What is Prewarping? Why is it needed? Nov/Dec 2008

In IIR design using bilinear transformation the conversion of specified digital frequencies to analog frequencies is called Pre-warping. The Pre-Warping is necessary to eliminate the effect of warping on amplitude response.

9. How one can design digital filters from analog filters?

1) Map the desired digital filter specifications into those for an equivalent analog filter.
2) Derive the analog transfer function for the analog prototype.
3) Transform the transfer function of the analog prototype into an equivalent digital filter transfer function.

10. Mention the procedures for digitizing the transfer function of an analog filter.

The two important procedures for digitizing the transfer function of an analog filter are

Impulse invariance method.

Bilinear transformation method.

11. What do you understand by backward difference?

One of the simplest method for converting an analog filter into a digital filter is to approximate the differential equation by an equivalent difference equation. d/dt y(t)=y(nT)-y(nT-T)/T

The above equation is called backward difference equation.

12. What is the mapping procedure between S-plane & Z-plane in the method of mapping differentials? What are its characteristics?

The mapping procedure between S-plane & Z-plane in the method of mapping of differentials is given by

$$H(Z) = H(S)|S=(1-Z^{-1})/T$$

The above mapping has the following characteristics

The left half of S-plane maps inside a circle of radius ½ centered at Z= ½ in the Z plane.

The right half of S-plane maps into the region outside the circle of radius ½ in the Z-plane.

The j-axis maps onto the perimeter of the circle of radius ½ in the Z-plane.

13. What is meant by impulse invariant method of designing IIR filter?

In this method of digitizing an analog filter, the impulse response of resulting digital filter is a sampled version of the impulse response of the analog filter.

14. Give the bilinear transform equation between S-plane & Z-plane.

$$S=2/T(1-Z^{-1}/1+Z^{-1})$$

15. What is bilinear transformation?

The bilinear transformation is a mapping that transforms the left half of S-plane into the unit circle in the Z-plane only once, thus avoiding aliasing of frequency components.The mapping from the S-plane to the Z-plane is in bilinear transformation is

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

16. What are the properties of bilinear transformation?

1) The mapping for the bilinear transformation is a one-to-one mapping that is for every point Z, there is exactly one corresponding point S, and vice-versa.
2) The j .-axis maps on to the unit circle |z|=1,the left half of the s-plane maps to the interior of the unit circle |z|=1 and the half of the s-plane maps on to the exterior of the unit circle |z|=1.

17. What are the advantages & disadvantages of bilinear transformation?

Advantages:

The bilinear transformation provides one-to-one mapping.
Stable continuous systems can be mapped into realizable, stable digital systems.

There is no aliasing.

Disadvantage:

The mapping is highly non-linear producing frequency, compression at high frequencies.

Neither the impulse response nor the phase response of the analog filter is preserved in a digital filter obtained by bilinear transformation.

18. Define signal flow graph.

A signal flow graph is a graphical representation of the relationships between the variables of a set of linear difference equations.

PART-B

1) **Derive the equation for designing IIR filter using bilinear transformation.**

ANS: Ref: Page No.336….Digital Signal Processing by A.Nagoorkani

2) **Derive the equation for designing IIR filter using impulse invariant method.**

ANS: Ref: Page No .330….Digital Signal Processing by A.Nagoorkani

3) **Draw all the possible realization of FIR system**.

Ref : Page No 502 Digital Signal Processing by John G.Proakis

4) **Design a digital Butterworth filter satisfying the constraints**

$$0.707 \le |H(\omega)| \le 1.0 \;;\; 0 \le \omega \le \pi/2$$

$$|H(\omega)| \le 0.2 \;;\; 3\pi/4 \le \omega \le \pi.$$

Nov/Dec 2006

Ans: Ref Pg.No 435-437, DSP by Salivahanan.

5) **Design a digital Butterworth filter satisfying the constraints**

$$0.8 \le |H(\omega)| \le 1.0 \;;\; 0 \le \omega \le \pi/4$$

$$|H(\omega)| \le 0.2 \;;\; \pi/2 \le \omega \le \pi.$$

May/June2007 & Nov/Dec 2008

Ans: Ref: Pg.No: 359-362, DSP by Nagoorkani.

6) **Design a digital BUTTERWORTH filter that satisfies the following constraint using BILINEAR Transformation. Assume T = 1 sec.**

$$0.9 \le |H(\omega)| \le 1 \;;\; 0 \le \omega \le \pi/2$$

$$|H(\omega)| \le 0.2 \;;\; (3\pi/4) \le \omega \le \pi$$

ANS: Ref: Page No .370….Digital Signal Processing by A.Nagoorkani

7) **For the analog transfer function H(S) = 2/ (S+1)(S+2) . Determine H(Z) using impulse invariant technique.**

April /May 2008

ANS: Ref: Page No .341.Digital Signal Processing by A.Nagoorkani

UNIT IV -FIR FILTER DESIGN

PART-A

1) Give any two properties of Butterworth and Chebyshev filter.
Nov/Dec 2006

Properties of Butterworth:

The butterworth filters are all pole design.
The filter order N completely specifies the filter
The magnitude is maximally flat at the origin.
The magnitude is monotomically decreasing function of ohm.

Properties of Chebyshev:

The magnitude reponse of the filter exhibits ripples in the pass band or stop band
The pole of the filter lies on an ellipse.

2) Show that the filter with h(n) = [-1,0,1] is a linear phase filter.
May /June 2007 & Nov/Dec 2008

Solution:

h(n) = [-1,0,1]
h(0) = -1 = -h(N-1-n) = -h(3-1-0) = -h(2)
h(1) = 0 = -h(N-1-n) = -h(3-1-1) = -h(1)
h(2) = 1 = -h(N-1-n) = -h(3-1-2) = -h(0)
It is a linear phase filter.

3) In the design of FIR digital filter, how is Kaiser Window different from other windows? **Nov/Dec 2007**

In all other windows a trade off exists between ripple ratio and main lobe width. In Kaiser Window both ripple ratio and main lobe width can be varied independently.

4) What are the merits and demerits of FIR filter? April/May 2008

Merits :

Linear phase filter.
Always Stable

Demerits:

The duration of the impulse response should be large
Non integral delay.

5) What are the advantages and disadvantages of FIR filters?

Advantages:

1. FIR filters have exact linear phase.
2. FIR filters are always stable.
3. FIR filters can be realized in both recursive and non recursive structure.
4. Filters with any arbitrary magnitude response can be tackled using FIR sequence.

Disadvantages:

1. For the same filter specifications the order of FIR filter design can be as high as 5 to 10 times that in an IIR design.
2. Large storage requirement is requirement
3. Powerful computational facilities required for the implementation.

6) What are the desirable characteristics of the window function?

The desirable characteristics of the window are

1. The central lobe of the frequency response of the window should contain most of the energy and should be narrow.
2. The highest side lobe level of the frequency response should be small.
3. The side lobes of the frequency response should decrease in energy

7) Mention the windows of FIR filters.

a. Rectangular window
b. Hamming window
c. Hanning window
d. Bartlett window
e. Kaiser window

8) What is the necessary and sufficient condition for linear phase characteristic in FIR filter?

The necessary and sufficient condition for linear phase characteristic in FIR filter is, the impulse response h(n) of the system should have the symmetry property i.e.,H(n) = h(N-1-n)where N is the duration of the sequence.

9) What are the advantages of Kaiser Window?

It provides flexibility for the designer to select the side lobe level
It has the attractive property that the side lobe level can be varied continuously from the low value in the Blackman window to the high value in the rectangular window.

10) What is the principle of designing FIR filter using frequency sampling method?

In frequency sampling method the desired magnitude response is sampled and a linear phase response is specified .The samples of desiredfrequency response are identified as DFT coefficients. The filter coefficients are then determined as the IDFT of this set of samples.

11) For what type of filters frequency sampling method is suitable?

Frequency sampling method is attractive for narrow band frequency selective filters where only a few of the samples of the frequency response are non zero.

12) When cascade form realization is preferred in FIR filters?

The cascade form realization is preferred when complex zeros with absolute magnitude is less than one.

13) Compare Rectangular & Hamming window.

S.No	Rectangular Window	Hamming window.
1.	The width of the main lobe in window spectrum is $4\pi/N$	The width of the main lobe in window spectrum is $8\pi/N$
2.	The maximun side lobe magnitude in window spectrum is -13 dB	The maximun side lobe magnitude in window spectrum is -41 dB

14) Compare Hamming window & Kaiser Window.

S.No	Kaiser Window	Hamming window.
1.	The width of the main lobe in window spectrum depends on the value of α and N.	The width of the main lobe in window spectrum is $8\pi/N$
2.	The maximun side lobe	The maximun side lobe

	magnitude with respect to peak of main lobe is variable using the parameter α.	magnitude in window spectrum is -41 dB

15) List the steps involved in the design of FIR filters using windows.

1. For the desired frequency response Hd(w), find the impulse response $h_d(n)$ using Equation

hd(n)=1/2 Hd(w)ejwndw

2 .Multiply the infinite impulse response with a chosen window sequence w(n) of length N to obtain filter coefficients h(n),i.e.,

h(n)= hd(n)w(n) for |n|._1-1)/2

= 0 otherwise

Find the transfer function of the realizable filter

(N-1)/2

ANS: Reference : Page No .294….Digital Signal Processing by A.Nagoorkani [First Edition]

16) **What is meant by limit cycle oscillation in digital filter**?

May /June 2007 & Nov/Dec 2007 &April/May 2008

In recursive system when the input is zero or same non-zero constant value the non linearities due to finite precision arithmetic operation may cause periodic oscillation in theoutput. Thus the oscillation is called as Limit cycle.

17) Express the fraction 7/8 and – 7/8 in sign magnitude, 2's complement and 1's complement. Nov/Dec 2006

Solution:

7/8 = 0.875 = $(0.111)_2$ is sign magnitude

1's Complement = $(0.111)_2$

2's Complement = $(0.111)_2$

7/8 = -0.875

Sign magnitude: $(1.111)_2$

1's Complement = $(1.000)_2$

2's Complement = $(1.001)_2$

18) Identify the various factors which degrade the performance of the digital filter implementation when finite word length is used. May /June 2007 & April/May 2008 & Nov/Dec 2008

Input quantization error
Coefficient quantization error
Product quantization error

19) a) What are the quantization errors due to finite word length register in digital filter.
b) What are the different quantization methods? Nov/Dec 2006

<u>Quantization Error :</u>

Input quantization error
Coefficient quantization error
Product quantization error

<u>Quantization methods</u>

Truncation
Rounding

20) Express the fraction (-7/32) in signed magnitude and 2's complement notations using 6 bits. Nov/Dec 2007 &Nov/Dec 2008

In Signed Magnitude: 1.001110
In 2's complement: 1.110010

PART-B

1. a) Design a high pass filter hamming window by taking 9 samples of w(n) and with a cutoff frequency of 1.2 radians/sec Nov/Dec 2006

Ans: Ref: Pg.No: 298-301, DSP by Nagoorkani.

2. a) Describe the design of FIR filter using frequency sampling technique.
b) The desired frequency response of a low pass filter is given by

$$H_d(\omega) = \begin{cases} e^{-j2\omega} & ; -\pi/4 \le \omega \le \pi/4 \\ 0 & ; \text{other wise.} \end{cases}$$

3. Obtain the filter coefficient, h(n) using RECTANGUAR window define by

$$W(n) = \begin{cases} 1; & 0 \le n \le 4 \\ 0; & \text{otherwise.} \end{cases}$$

Nov/Dec 2007

Ans: a) Ref Pg.No 389-391, DSP by Salivahanan.
b) Ref Pg.No 399, DSP by Salivahanan.

4. ii) Determine the magnitude response of the FIR filter (M=11) and show that phase and group delay are conatant.

iii) The desired frequency response of a low pass filter is given by

$$H_d(\omega) = \begin{cases} e^{-j3\omega} & ; -3\pi/4 \le \omega \le 3\pi/4 \\ 0 & ; \text{other wise.} \end{cases}$$

Determine $H(e^{j\omega})$ for M= 7using HAMMING window.

Ans: a) i) Ref Pg.No 437-439, DSP by Salivahanan.
ii) Ref Pg.No 383-384, DSP by Salivahanan.
iii) Ref Pg.No 400-401, DSP by Salivahanan.
iv) Ref Pg.No 426, DSP by Salivahanan.

5. Explain the design of lowpass digital butterworth filter.

ANS: Reference : Page No.347….Digital Signal Processing by A.Nagoorkani [First Edition]

6. Explain the design of lowpass digital Chebyshev filter.

ANS: Reference : Page No.351….Digital Signal Processing by A.Nagoorkani [First Edition]

7. Derive the frequency response of linear phase FIR filter when impulse response is Symmetric when N is ODD.

ANS: Reference : Page No.264….Digital Signal Processing by A.Nagoorkani [First Edition]

8. Derive the frequency response of linear phase FIR filter when impulse response is Symmetric when N is EVEN.

ANS: Reference : Page No.267….Digital Signal Processing by A.Nagoorkani [First Edition]

9. Derive the frequency response of linear phase FIR filter when impulse response is Anti symmetric when N is ODD.

ANS: Reference : Page No.269….Digital Signal Processing by A.Nagoorkani [First Edition]

10.Derive the frequency response of linear phase FIR filter when impulse response is Antisymmetric when N is EVEN.

ANS: Reference : Page No.272….Digital Signal Processing by A.Nagoorkani [First Edition]

11.Analyze the Windows specifying the FIR filters

a. Rectangular window
b. Hamming window
c. Hanning window
d. Bartlett window
e. Kaiser window

ANS: Reference : Page No .281….Digital Signal Processing by A.Nagoorkani [First Edition]

12.Draw the structures of FIR filters

Reference : Page No 502Digital Signal Processing by John G.Proakis[Third Edition]

13..Explain the design of FIR filters by frequency sampling technique.

ANS: Reference : Page No.306….Digital Signal Processing by A.Nagoorkani [First Edition]

14.a)i) Explain the characteristics of a Limit cycle oscillation w.r.t the system described by the difference equation $y(n) = 0.95y(n-1)+x(n)$.Determine the dead band of the filter.

ii) Draw the product quantisation noise model of second order IIR fihter. Nov/Dec 2006 & Nov/Dec 2007

Ans: a) i) Dead band = [+0.625.-0.625]

ii) Ref Pg.No 513-514, DSP by Salivahanan.

15.For the given transfer function $H(Z) = H_1(Z) .H_2(Z)$,where

$$H_1(Z) = \frac{1}{1 - 0.5\,Z^{-1}} \quad \text{and } H_2(Z) = \frac{1}{1 - 0.6\,Z^{-1}}$$

Find the output round off noise power. **Nov/Dec 2006**

Ans: $2^{-2b}/12(5.4315)$

16.a)i) Consider the truncation of negative fraction number represented in(β+1) bit fixed point binary form including sign bit . Let (β-b) bits be truncated .Obtain the range of truncation errors for signed magnitude ,2's complement and 1's complement representation of negative numbers. **Nov/Dec 2007**

ii) The coefficients of a system defined by

$$H(Z) = \frac{1}{(1-0.4Z^{-1})(1-0.55Z^{-1})}$$

are represented in anumber with a sign bit and 3 data bits.

iii) Consider the (b+1) bit bipolar A/D converter.Obtain an expression for signal to quantization noise ratio .

May /June 2007& Nov/Dec 2007&April/May2008 & Nov/Dec 2008

Ans: a) i) Ref Pg.No 496-499, DSP by Salivahanan.

ii) Direct form: $1/ [1-0.875z^{-1}+0.125Z^{-2}]$

Cascade form: $1/[1-0.375Z^{-1}][1-0.5Z^{-1}]$

17. With the neat diagram explain the operation of limit cycle oscillations.

Ref Pg.No 513-514, DSP by Salivahanan.

UNIT V - APPLICATIONS
PART-A

1. Define multirate digital signal processing.

The process of converting a signal from a given rate to a different rate is called sampling rate conversion. The system that employs multiple sampling rates in the processing of digital signals are called digital signal processing systems.

2. Give the advantages of multirate digital signal processing.

- Computational requirements are less
- Storage for filter coefficients is less
- Finite arithmetic effects are less
- Sensitivity to filter coefficients lengths are less

3. **Give the applications of multirate digital signal processing.**
    - Communication systems
    - Speech and audio processing systems
    - Antenna systems
    - Radar systems

4. **Define Decimation.**
   The process of reducing the sampling rate of the signal is called decimation (sampling rate compression).

5. **Define Interpolation**
   The process of increasing the sampling rate of the signal is called interpolation (sampling rate Expansion).

PART-B

1. **Explain briefly: Multi rate signal processing** May/June 2007
   Ref Pg.No 751, DSP by Proakis
2. **Explain briefly: Vocoder** May/June 2007
   Ref Pg.No 754, DSP by Proakis
3. **Explain decimation of sampling rate by an integer factor D and derive spectra for decimated signal** **May/June 2006**
   Ref Pg.No 755, DSP by Proakis
4. **Explain interpolation of sampling rate by an integer factor I and derive spectra for decimated signal** **May/June 2006**
   Ref Pg.No 760, DSP by Proakis
5. **Explain about adaptive filters**
   Ref Pg.No 880, DSP by Proakis