5.1 SYLLABUS

EC2312 DIGITAL SIGNAL PROCESSING 3 1 0 4

1. INTRODUCTION 9

Classification of systems: Continuous, discrete, linear, causal, stable, dynamic, recursive, time variance; classification of signals: continuous and discrete, energy and power; athematical representation of signals; spectral density; sampling techniques, quantization, quantization error, Nyquist rate, aliasing effect. Digital signal representation.

2. DISCRETE TIME SYSTEM ANALYSIS 9

Z-transform and its properties, inverse z-transforms; difference equation – Solution by z-transform, application to discrete systems - Stability analysis, frequency response – onvolution – Fourier transform of discrete sequence – Discrete Fourier series.

3. DISCRETE FOURIER TRANSFORM & COMPUTATION 9

DFT properties, magnitude and phase representation - Computation of DFT using FFT algorithm – DIT & DIF - FFT using radix 2 – Butterfly structure.

4. DESIGN OF DIGITAL FILTERS 9

FIR & IIR filter realization – Parallel & cascade forms. FIR design: Windowing Techniques – Need and choice of windows – Linear phase characteristics. IIR design: Analog filter design - Butterworth and Chebyshev approximations; digital design using impulse invariant and bilinear transformation - Warping, prewarping - Frequency transformation.

5. DIGITAL SIGNAL PROCESSORS 9

Introduction – Architecture – Features – Addressing Formats – Functional modes - Introduction to Commercial Processors

TOTAL : 45 PERIODS

TEXT BOOKS

1. J.G. Proakis and D.G. Manolakis, 'Digital Signal Processing Principles, Algorithms and Applications', Pearson Education, New Delhi, 2003 / PHI.

2. S.K. Mitra, 'Digital Signal Processing – A Computer Based Approach', Tata McGraw Hill, New Delhi, 2001.

5.2 SHORT QUESTIONS AND ANSWERS

UNIT-I - SIGNALS & SYSTEMS

1. Define Signal.

A Signal is defined as any physical quantity that varies with time, space or any other independent variables.

2. Define a system.

A System is a physical device (i.e., hardware) or algorithm (i.e., software) that performs an operation on the signal.

3. What are the steps involved in digital signal processing?

- Converting the analog signal to digital signal, this is performed by A/D converter
- Processing Digital signal by digital system.
- Converting the digital signal to analog signal, this is performed by D/A converter.

4. **Give some applications of DSP?**

- Speech processing – Speech compression & decompression for voice storage system
- Communication – Elimination of noise by filtering and echo cancellation.
- Bio-Medical – Spectrum analysis of ECG,EEG etc.

5. Write the classifications of DT Signals.

- Energy & Power signals
- Periodic & Aperiodic signals
- Even & Odd signals.

6. What is an Energy and Power signal?

Energy signal:
A finite energy signal is periodic sequence, which has a finite energy but zero average power.
Power signal:
An Infinite energy signal with finite average power is called a power signal.

7. What is Discrete Time Systems?

The function of discrete time systems is to process a given input sequence to generate output sequence. In practical discrete time systems, all signals are digital signals, and operations on such signals also lead to digital signals. Such discrete time systems are called digital filter.

8. Write the Various classifications of Discrete-Time systems.

- Linear & Non linear system
- Causal & Non Causal system
- Stable & Un stable system
- Static & Dynamic systems

9. Define Linear system

A system is said to be linear system if it satisfies Super position principle. Let us consider x1(n) & x2(n) be the two input sequences & y1(n) & y2(n) are the responses respectively,

$$T[ax1(n) + bx2(n)] = a\ y1(n) + by2(n)$$

10. Define Static & Dynamic systems

When the output of the system depends only upon the present input sample, then it is called static system, otherwise if the system depends past values of input then it is called dynamic system

11. Define causal system.

When the output of the system depends only upon the present and past input sample, then it is called causal system, otherwise if the system depends on future values of input then it is called non-causal system

12. Define Shift-Invariant system.

If y(n) is the response to an input x(n), then the response to an input

$$X(n) = x(n-n0) \text{ then } \quad y(n) = y(n-n0)$$

When the system satisfies above condition then it is said to shift in variant, otherwise it is variant.

13. Define impulse and unit step signal.

Impulse signal $\delta(n)$:

The impulse signal is defined as a signal having unit magnitude at n = 0 and zero for other values of n.

$$\delta(n) = 1;\ n = 0$$
$$0;\ n \neq 0$$

Unit step signal u(n):

The unit step signal is defined as a signal having unit magnitude for all values of $n \geq 0$

$$u(n) = 1;\ n \geq 0$$
$$0;\ n < 0$$

14. What are FIR and IIR systems?

The impulse response of a system consist of infinite number of samples are called IIR system & the impulse response of a system consist of finite number of samples are called FIR system.

15. What are the basic elements used to construct the block diagram of discrete time system?

The basic elements used to construct the block diagram of discrete time Systems are Adder, Constant multiplier &Unit delay element.

16. What is ROC in Z-Transform?

The values of z for which z – transform converges is called region of convergence (ROC). The z-transform has an infinite power series; hence it is necessary to mention the ROC along with z-transform.

17. List any four properties of Z-Transform.

- Linearity
- Time Shifting
- Frequency shift or Frequency translation
- Time reversal

18. What are the different methods of evaluating inverse z-transform?

- Partial fraction expansion
- Power series expansion
- Contour integration (Residue method)

19. Define sampling theorem.

A continuous time signal can be represented in its samples and recovered back if the sampling frequency $F_s \geq 2B$. Here 'F_s' is the sampling frequency and 'B' is the maximum frequency present in the signal.

20. Check the linearity and stability of g(n),

- Since square root is nonlinear, the system is nonlinear.
- As long as x(n) is bounded, its square root is bounded. Hence this system is stable.

21. What are the properties of convolution?

1. Commutative property $x(n) * h(n) = h(n) * x(n)$
2. Associative property $[x(n) * h_1(n)]*h_2(n) = x(n)*[h_1(n) * h_2(n)]$
3. Distributive property $x(n) *[h_1(n)+h_2(n)] = [x(n)*h_1(n)]+[x(n) * h_2(n)]$

UNIT-II
DISCRETE TIME SYSTEM ANALYSIS

1. Define DTFT.

Let us consider the discrete time signal x(n).Its DTFT is denoted as X(w).It is given as
$X(w)= x(n)e^{-jwn}$

2. State the condition for existence of DTFT?

The conditions are
• If x(n)is absolutely summable then
|x(n)|<
• If x(n) is not absolutely summable then it should have finite energy for
DTFT to exit.

3. List the properties of DTFT.

Periodicity
Linearity
Time shift
Frequency shift
Scaling
Differentiation in frequency domain
Time reversal
Convolution
Multiplication in time domain
Parseval's theorem

4. What is the DTFT of unit sample?

The DTFT of unit sample is 1 for all values of w.

5. Define DFT.
DFT is defined as X(w)= x(n)e-jwn.
Here x(n) is the discrete time sequence
X(w) is the fourier transform ofx(n).

6. Define Twiddle factor.

The Twiddle factor is defined as $WN=e^{-j2 /N}$

7. Define Zero padding.

The method of appending zero in the given sequence is called as Zero padding.

8. Define circularly even sequence.

A Sequence is said to be circularly even if it is symmetric about the point zero on
the circle. x(N-n)=x(n),1<=n<=N-1.

9. Define circularly odd sequence.

A Sequence is said to be circularly odd if it is anti symmetric about point x(0) on the circle

10. Define circularly folded sequences.

A circularly folded sequence is represented as x((-n))N. It is obtained by plotting x(n) in clockwise direction along the circle.

11. State circular convolution.

This property states that multiplication of two DFT is equal to circular convolution of their sequence in time domain.

12. State parseval's theorem.

Consider the complex valued sequences x(n) and y(n).If
x(n)y*(n)=1/N X(k)Y*(k)

13. Define Z transform.

The Z transform of a discrete time signal x(n) is denoted by X(z) and is given
by $X(z)= x(n)Z^{-n.}$

14. Define ROC.

The value of Z for which the Z transform converged is called region of convergence.

15. Find Z transform of x(n)={1,2,3,4}

x(n)= {1,2,3,4}
$X(z)= x(n)z^{-n}$
$= 1+2z^{-1}+3z^{-2}+4z^{-3.}$
$= 1+2/z+3/z^{2}+4/z^{3}.$

16. State the convolution property of Z transform.

The convolution property states that the convolution of two sequences in time domain is equivalent to multiplication of their Z transforms.

17. What z transform of (n-m)?

By time shifting property
$Z[A (n-m)]=AZ^{-m}$ sin Z[(n)] =1

18. State initial value theorem.

If x(n) is causal sequence then its initial value is given by x(0)=lim X(z)

19. List the methods of obtaining inverse Z transform.
Inverse z transform can be obtained by using
Partial fraction expansion.

Contour integration
Power series expansion
Convolution.

20. Obtain the inverse z transform of X(z)=1/z-a,|z|>|a|

Given X(z)=z-1/1-az-1
By time shifting property
X(n)=an.u(n-1)

UNIT-III - DISCRETE FOURIER TRANSFORM AND COMPUTATION

1. What is DFT?

It is a finite duration discrete frequency sequence, which is obtained by sampling one period of Fourier transform. Sampling is done at N equally spaced points over the period extending from w=0 to 2л.

2. Define N point DFT.

The DFT of discrete sequence x(n) is denoted by X(K). It is given by,
Here k=0,1,2…N-1
Since this summation is taken for N points, it is called as N-point DFT.

3. What is DFT of unit impulse δ(n)?

The DFT of unit impulse δ(n) is unity.

4. List the properties of DFT.

Linearity, Periodicity, Circular symmetry, symmetry, Time shift, Frequency shift, complex conjugate, convolution, correlation and Parseval's theorem.

5. State Linearity property of DFT.

DFT of linear combination of two or more signals is equal to the sum of linear combination of DFT of individual signal.

6. When a sequence is called circularly even?

The N point discrete time sequence is circularly even if it is symmetric about the point zero on the circle.

7. What is the condition of a sequence to be circularly odd?

An N point sequence is called circularly odd it if is antisymmetric about point zero on the circle.

8. Why the result of circular and linear convolution is not same?

Circular convolution contains same number of samples as that of x (n) and h (n), while in linear convolution, number of samples in the result (N) are,

$$N=L+M-1$$

Where L= Number of samples in x (n)
M=Number of samples in h (n)

9. What is circular time shift of sequence?

Shifting the sequence in time domain by '1' samples is equivalent to multiplying the sequence in frequency domain by W_N^{kl}

10. **What is the disadvantage of direct computation of DFT?**

For the computation of N-point DFT, N2 complex multiplications and N[N-1] Complex additions are required. If the value of N is large than the number of computations will go into lakhs. This proves inefficiency of direct DFT computation.

11. What is the way to reduce number of arithmetic operations during DFT computation?

Number of arithmetic operations involved in the computation of DFT is greatly reduced by using different FFT algorithms as follows.

1. Radix-2 FFT algorithms.

-Radix-2 Decimation in Time (DIT) algorithm.

- Radix-2 Decimation in Frequency (DIF) algorithm.

2. Radix-4 FFT algorithm.

12. What is the computational complexity using FFT algorithm?

1. Complex multiplications = $N/2 \log^2 N$
2. Complex additions = $N \log^2 N$

13. How linear filtering is done using FFT?

Correlation is the basic process of doing linear filtering using FFT. The correlation is nothing but the convolution with one of the sequence, folded. Thus, by folding the sequence h (n), we can compute the linear filtering using FFT.

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

Let the sequence x (n) has a length L. If we want to find the N point DFT (N>L) of the sequence x (n). This is known as zero padding. The uses of padding a sequence with zeros are

(i) We can get 'better display' of the frequency spectrum.
(ii) With zero padding, the DFT can be used in linear filtering.

15. Why FFT is needed?

The direct evaluation of the DFT using the formula requires N^2 complex multiplications and N (N-1) complex additions. Thus for reasonably large values of N (inorder of 1000) direct evaluation of the DFT requires an inordinate amount of computation. By using FFT algorithms the number of computations can be reduced. For example, for an N-point DFT, The number of complex multiplications required using FFT is $N/2\log_2 N$. If N=16, the number of complex multiplications required for direct evaluation of DFT is 256, whereas using DFT only 32 multiplications are required.

16. What is the speed of improvement factor in calculating 64-point DFT of a sequence using direct computation and computation and FFT algorithms?
Or
Calculate the number of multiplications needed in the calculation of DFT and FFT with 64-point sequence.

The number of complex multiplications required using direct computation is
N^2=642=4096.
The number of complex multiplications required using FFT is
$N/2 \log_2 N = 64/2\log_2 64=192$.
Speed improvement factor = 4096/192=21.33

17. What is the main advantage of FFT?

FFT reduces the computation time required to compute discrete Fourier transform.

18. Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with using FFT algorithm with 32-point sequence.

For N-point DFT the number of complex multiplications needed using FFT algorithm is $N/2 \log_2 N$.
For N=32, the number of the complex multiplications is equal to $32/2\log_2 32=16*5=80$.

19. What is FFT?

The fast Fourier transforms (FFT) is an algorithm used to compute the DFT. It makes use of the Symmetry and periodically properties of twiddles factor W^K_N
to effectively reduce the DFT computation time. It is based on the fundamental principle of decomposing the computation of the DFT of a sequence of length N into successively smaller discrete Fourier transforms. The FFT algorithm provides speed-increase factors, when compared with direct computation of the DFT, of approximately 64 and 205 for 256-point and 1024-point transforms, respectively.

20. How many multiplications and additions are required to compute N-point DFT using redix-2 FFT?

The number of multiplications and additions required to compute N-point DFT using redix-2 FFT are $N \log_2 N$ and $N/2 \log_2 N$ respectively.

21. What is meant by radix-2 FFT?
The FFT algorithm is most efficient in calculating N-point DFT. If the number of output points N can be expressed as a power of 2, that is, $N=2^M$, where M is an integer, Then this algorithm is known as radix-s FFT algorithm

22. What is a decimation-in-time algorithm?

Decimation-in-time algorithm is used to calculate the DFT of a N-point Sequence. The idea is to break the N-point sequence into two sequences, the DFTs of which can be combined to give the DFT of the original N-point sequence. Initially the N-point sequence is divided into two N/2-point sequences $x_e(n)$ and $x_0(n)$, which have the even and odd members of x(n) respectively. The N/2 point DFTs of these two sequences are evaluated and combined to give the N point DFT. Similarly the N/2 point DFTs can be expressed as a combination of N/4 point DFTs. This process is continued till we left with 2-point DFT. This algorithm is called Decimation-in-time because the sequence x(n) is often splitted into smaller sub sequences.

23. What are the differences and similarities between DIF and DIT algorithms?
Differences:

1. For DIT, the input is bit reversal while the output is in natural order, whereas for DIF, the input is in natural order while the output is bit reversed.
2. The DIF butterfly is slightly different from the DIT butterfly, the difference being that the complex multiplication takes place after the add-subtract operation in DIF.

Similarities: Both algorithms require same number of operations to compute the DFT. Bot algorithms can be done in place and both need to perform bit reversal at some place during the computation.

24. What are the applications of FFT algorithms?

1. Linear filtering
2. Correlation
3. Spectrum analysis

25. What is a decimation-in-frequency algorithm?

In this the output sequence X (K) is divided into two N/2 point sequences and each N/2 point sequences are in turn divided into two N/4 point sequences.

26. Distinguish between DFT and DTFT.

S.No	DFT	DTFT
1.	Obtained by performing sampling operation in both the time and frequency domains.	Sampling is performed only in time domain.
2.	Discrete frequency spectrum	Continuous function of ω

27. Distinguish between Fourier series and Fourier transform.

S.No.	Fourier Series	Fourier transform
1	Gives the harmonic content of a periodic time function.	Gives the frequency information for an aperiodic signal.
2.	Discrete frequency spectrum	Continuous frequency spectrum

UNIT-IV - DESIGN OF DEGITAL FILTER

1) Define IIR filter?

IIR filter has Infinite Impulse Response.

2) What are the various methods to design IIR filters?

- Approximation of derivatives
- Impulse invariance
- Bilinear transformation.

3) Which of the methods do you prefer for designing IIR filters? Why?

Bilinear transformation is best method to design IIR filter, since there is no aliasing in it.

4) What is the main problem of bilinear transformation?

Frequency warping or nonlinear relationship is the main problem of bilinear transformation.

5) What is prewarping?

Prewarping is the method of introducing nonlinearly in frequency relationship to compensate warping effect.

6) State the frequency relationship in bilinear transformation?

$$\Omega = \frac{2}{T} \tan (w/2)$$

7) Where the jΩ axis of s-plane is mapped in z-plane in bilinear transformation?

The jΩ axis of s-plane is mapped on the unit circle in z-plane in bilinear transformation

8) Where left hand side and right hand side are mapped in z-plane in bilinear transformation?

Left hand side -- Inside unit circle
Right hand side – Outside unit circle

9) What is the frequency response of Butterworth filter?

Butterworth filter has monotonically reducing frequency response.

10) Which filter approximation has ripples in its response?

Chebyshev approximation has ripples in its pass band or stop band.

11) Can IIR filter be designed without analog filters?

Yes. IIR filter can be designed using pole-zero plot without analog filters

12) What is the advantage of designing IIR Filters using pole-zero plots?

The frequency response can be located exactly with the help of poles and zeros.

13) Compare the digital and analog filter.

Digital filter	Analog filter
i) Operates on digital samples of the signal. ii) It is governed by linear difference equation. iii) It consists of adders, multipliers and delays implemented in digital logic. iv) In digital filters the filter coefficients are designed to satisfy the desired frequency response.	i) Operates on analog signals. ii) It is governed by linear difference equation. iii) It consists of electrical components like resistors, capacitors and inductors. iv) In digital filters the approximation problem is solved to satisfy the desired frequency response.

14) What are the advantages and disadvantages of digital filters?

<u>Advantages of digital filters</u>

- High thermal stability due to absence of resistors, inductors and capacitors.
- Increasing the length of the registers can enhance the performance characteristics like accuracy, dynamic range, stability and tolerance.
- The digital filters are programmable.
- Multiplexing and adaptive filtering are possible.

<u>Disadvantages of digital filters</u>

- The bandwidth of the discrete signal is limited by the sampling frequency.
- The performance of the digital filter depends on the hardware used to implement the filter.

15) What is impulse invariant transformation?

The transformation of analog filter to digital filter without modifying the impulse response of the filter is called impulse invariant transformation.

16) Obtain the impulse response of digital filter to correspond to an analog filter with impulse response $h_a(t) = 0.5\, e^{-2t}$ and with a sampling rate of 1.0kHz using impulse invariant method.

17) How analog poles are mapped to digital poles in impulse invariant transformation?

In impulse invariant transformation the mapping of analog to digital poles are as follows,

- The analog poles on the left half of s-plane are mapped into the interior of unit circle in z-plane.
- The analog poles on the imaginary axis of s-plane are mapped into the unit circle in the z-plane.
- The analog poles on the right half of s-plane are mapped into the exterior of unit circle in z-plane.

18) What is the importance of poles in filter design?

The stability of a filter is related to the location of the poles. For a stable analog filter the poles should lie on the left half of s-plane. For a stable digital filter the poles should lie inside the unit circle in the z-plane.

19) Why an impulse invariant transformation is not considered to be one-to-one?

In impulse invariant transformation any strip of width $2\pi/T$ in the s-plane for values of s-plane in the range $(2k-1)/T \leq \Omega \leq (2k-1)\,\pi/T$ is mapped into the entire z-plane. The left half of each strip in s-plane is mapped into the interior of unit circle in z-plane, right half of each strip in s-plane is mapped into the exterior of unit circle in z-plane and the imaginary axis of each strip in s-plane is mapped on the unit circle in z-plane. Hence the impulse invariant transformation is many-to-one.

20) What is Bilinear transformation?

The bilinear transformation is conformal mapping that transforms the s-plane to z-plane. In this mapping the imaginary axis of s-plane is mapped into the unit circle in z-plane, The left half of s-plane is mapped into interior of unit circle in z-plane and the right half of s-plane is mapped into exterior of unit circle in z-plane. The Bilinear mapping is a one-to-one mapping and it is accomplished when

21) How the order of the filter affects the frequency response of Butterworth filter.

The magnitude response of butterworth filter is shown in figure, from which it can be observed that the magnitude response approaches the ideal response as the order of the filter is increased.

22) Write the properties of Chebyshev type –1 filters.

- The magnitude response is equiripple in the passband and monotonic in the stopband.
- The chebyshev type-1 filters are all pole designs.
- The normalized magnitude function has a value of at the cutoff frequency Ω_c.
- The magnitude response approaches the ideal response as the value of N increases.

23) Compare the Butterworth and Chebyshev Type-1 filters.

Butterworth	Chebyshev Type - 1
i. All pole design. ii. The poles lie on a circle in s-plane. iii. The magnitude response is maximally flat at the origin and monotonically decreasing function of Ω. iv. The normalized magnitude response has a value of $1 / \sqrt{2}$ at the cutoff frequency Ω_c. v. Only few parameters has to be calculated to determine the transfer function.	i. All pole design. ii. The poles lie on a ellipse in s-plane. iii. The magnitude response is equiripple in passband and monotonically decreasing in the stopband. iv. The normalized magnitude response has a value of $1 / \sqrt{(1+\varepsilon^2)}$ at the cutoff frequency Ω_c. v. A large number of parameters has to be calculated to determine the transfer function.

22. What is FIR filters?

The specifications of the desired filter will be given in terms of ideal frequency response $H_d(w)$. The impulse response $h_d(n)$ of the desired filter can be obtained by inverse fourier transform of $H_d(w)$, which consists of infinite samples. The filters designed by selecting finite number of samples of impulse response are called FIR filters.

23. 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, and previous output samples.

24. What are the different types of filter based on frequency response?

The filters can be classified based on frequency response. They are I) Low pass filter ii) High pass filter iii) Band pass filter iv) Band reject filter.

25. Distinguish between FIR and IIR filters.

S.No.	FIR filter	IIR filter
1.	These filters can be easily designed to have perfectly linear phase.	These filters do not have linear phase.
2.	FIR filters can be realized recursively and non-recursively.	IIR filters can be realized recursively.
3.	Greater flexibility to control the shape of their magnitude response.	Less flexibility,usually limited to kind of filters.
4.	Errors due to roundoff noise are less severe in FIR filters, mainly because feedback is not used.	The roundoff noise in IIR filters are more.

26. What are the techniques of designing FIR filters?

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

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

A digital filter is causal if its impulse response h(n) = 0 for n<0
A digital filter is stable if its impulse response is absolutely summable,

28. What is the reason that FIR filter is always stable?

FIR filter is always stable because all its poles are at origin.

29. What are the properties of FIR filter?

1. FIR filter is always stable.
2. A realizable filter can always be obtained.
3. FIR filter has a linear phase response.
4.

30. How phase distortion and delay distortions are introduced?

The phase distortion is introduced when the phase characteristics of a filter is not linear within the desired frequency band.
The delay distortion is introduced when the delay is not constant within the desired frequency range.

31. Write the steps involved in FIR filter design.

- Choose the desired (ideal) frequency response $H_d(w)$.
- Take inverse fourier transform of $H_d(w)$ to get $h_d(n)$.
- Convert the infinite duration $h_d(n)$ to finite duration h(n).
- Take Z-transform of h(n) to get the transfer function H(z) of the FIR filter.

32. What are the advantages of FIR filters?

- Linear phase FIR filter can be easily designed.
- Efficient realization of FIR filter exist as both recursive and nonrecursive structures.
- FIR filters realized nonrecursively are always stable.
- The roundoff noise can be made small in nonrecursive realization of FIR filters.

33. What are the disadvantages of FIR filters?

- The duration of impulse response should be large to realize sharp cutoff filters.
- The non-integral delay can lead to problems in some signal processing applications.
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34. What is the necessary and sufficient condition for the linear phase characteristic of an FIR filter?

The necessary and sufficient condition for the linear phase characteristic of an FIR filter is that the phase function should be a linear function of w, which in turn requires constant phase and group delay.

35. What are the conditions to be satisfied for constant phase delay in linear phase FIR filters?

The conditions for constant phase delay ARE
Phase delay, $\alpha = (N-1)/2$ (i.e., phase delay is constant)
Impulse response, $h(n) = -h(N-1-n)$ (i.e., impulse response is antisymmetric)

36. How constant group delay & phase delay is achieved in linear phase FIR filters?

The following conditions have to be satisfied to achieve constant group delay & phase delay.
Phase delay, $\alpha = (N-1)/2$ (i.e., phase delay is constant)
Group delay, $\beta = \pi/2$ (i.e., group delay is constant)
Impulse response, $h(n) = -h(N-1-n)$ (i.e., impulse response is antisymmetric)

37. What are the possible types of impulse response for linear phase FIR filters?

There are four types of impulse response for linear phase FIR filters
- Symmetric impulse response when N is odd.
- Symmetric impulse response when N is even.
- Antisymmetric impulse response when N is odd.
- Antisymmetric impulse response when N is even.

38. List the well-known design techniques of linear phase FIR filters.

There are three well-known design techniques of linear phase FIR filters. They are
- Fourier series method and window method
- Frequency sampling method.
- Optimal filter design methods.

39. What is Gibb's phenomenon (or Gibb's Oscillation)?

In FIR filter design by Fourier series method the infinite duration impulse response is truncated to finite duration impulse response. The abrupt truncation of impulse response introduces oscillations in the passband and stopband. This effect is known as Gibb's phenomenon (or Gibb's Oscillation).

40. When cascade form realization is preferred in FIR filters?

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

41. What are the desirable characteristics of the frequency response of window function?

The desirable characteristics of the frequency response of window function are
- The width of the mainlobe should be small and it should contain as much of the total energy as possible.
- The sidelobes should decrease in energy rapidly as w tends to π.

42. Write the procedure for designing FIR filter using frequency-sampling method.

- Choose the desired (ideal) frequency response $H_d(w)$.
- Take N-samples of $H_d(w)$ to generate the sequence
- Take inverse DFT of to get the impulse response h(n).
- The transfer function H(z) of the filter is obtained by taking z-transform of impulse response.

43. What are the drawback in FIR filter design using windows and frequency sampling method? How it is overcome?

The FIR filter design using windows and frequency sampling method does not have Precise control over the critical frequencies such as w_p and w_s.
This drawback can be overcome by designing FIR filter using Chebyshev approximation technique.In this technique an error function is used to approximate the ideal frequency response, in order to satisfy the desired specifications.

44. Write the characteristic features of rectangular window.

- The mainlobe width is equal to $4\pi/N$.
- The maximum sidelobe magnitude is –13dB.
- The sidelobe magnitude does not decrease significantly with increasing w.

45. List the features of FIR filter designed using rectangular window.

- The width of the transition region is related to the width of the mainlobe of window spectrum.
- Gibb's oscillations are noticed in the passband and stopband.
- The attenuation in the stopband is constant and cannot be varied.

46. Why Gibb's oscillations are developed in rectangular window and how it can be eliminated or reduced?

The Gibb's oscillations in rectangular window are due to the sharp transitions from 1 to 0 at the edges of window sequence.
These oscillations can be eliminated or reduced by replacing the sharp transition by gradual transition. This is the motivation for development of triangular and cosine windows.

47. List the characteristics of FIR filters designed using windows.

- The width of the transition band depends on the type of window.
- The width of the transition band can be made narrow by increasing the value of N where N is the length of the window sequence.
- The attenuation in the stop band is fixed for a given window, except in case of Kaiser window where it is variable.

48. Compare the rectangular window and hanning window.

Rectangular window	Hanning Window
i) The width of mainlobe in window spectrum is $4\pi/N$ ii) The maximum sidelobe magnitude in window spectrum is –13dB. iii) In window spectrum the sidelobe magnitude slightly decreases with increasing w. iv) In FIR filter designed using rectangular window the minimum stopband attenuation is 22dB.	i)The width of mainlobe in window spectrum is $8\pi/N$ ii) The maximum sidelobe magnitude in window spectrum is –31dB. iii) In window spectrum the sidelobe magnitude decreases with increasing w. iv) In FIR filter designed using hanning window the minimum stopband attenuation is 44dB.

49. Compare the rectangular window and hamming window.

Rectangular window	Hamming Window
i) The width of mainlobe in window spectrum is $4\pi/N$ ii) The maximum sidelobe magnitude in window spectrum is –13dB. iii) In window spectrum the sidelobe magnitude slightly decreases with increasing w. iv) In FIR filter designed using rectangular window the minimum stopband attenuation is 22dB.	i)The width of mainlobe in window spectrum is $8\pi/N$ ii) The maximum sidelobe magnitude in window spectrum is –41dB. iii) In window spectrum the sidelobe magnitude remains constant. iv) In FIR filter designed using hamming window the minimum stopband attenuation is 44dB.

50. Write the characteristic features of hanning window spectrum.

- The mainlobe width is equal to $8\pi/N$.
- The maximum sidelobe magnitude is –41dB.
- The sidelobe magnitude remains constant for increasing w.

51. What is the mathematical problem involved in the design of window function?

The mathematical problem involved in the design of window function(or sequence) is that of finding a time-limited function whose Fourier Transform best approximates a band limited function. The approximation should be such that the maximum energy is confined to mainlobe for a given peak sidelobe amplitude.

52. List the desirable features of Kaiser Window spectrum.

- The width of the mainlobe and the peak sidelobe are variable.
- The parameter α in the Kaiser Window function is an independent variable that can be varied to control the sidelobe levels with respect to mainlobe peak.
- The width of the mainlobe in the window spectrum can be varied by varying the length N of the window sequence.

53. Compare the hamming window and Kaiser window.

Hamming Window	Kaiser Window
i)The width of mainlobe in window spectrum is $8\pi/N$ ii) The maximum sidelobe magnitude in window spectrum is –41dB. iii) In window spectrum the sidelobe magnitude remains constant. iv) In FIR filter designed using hamming window the minimum stopband attenuation is 44dB.	i) The width of mainlobe in window spectrum depends on the values of α & N. ii) The maximum sidelobe magnitude with respect to peak of mainlobe is variable using the parameter α. iii) In window spectrum the sidelobe magnitude decreases with increasing w. iv) In FIR filter designed using Kaiser window the minimum stopband attenuation is variable and depends on the value of α.

UNIT V - DIGITAL SIGNAL PROCESSOR

1. Write short notes on general purpose DSP processors

General-purpose digital signal processors are basically high speed microprocessors with hard ware architecture and instruction set optimized for DSP operations. These processors make extensive use of parallelism, Harvard architecture, pipelining and dedicated hardware whenever possible to perform time consuming operations
.

2. Write notes on special purpose DSP processors.

There are two types of special; purpose hardware.
(i) Hardware designed for efficient execution of specific DSP algorithms such as digital filter, FFT.
(ii) Hardware designed for specific applications, for example telecommunication, digital audio.

3. Briefly explain about Harvard architecture.

The principal feature of Harvard architecture is that the program and the data memories lie in two separate spaces, permitting full overlap of instruction fetch and execution.
Typically these types of instructions would involve their distinct type.
1. Instruction fetch
2. Instruction decode
3. Instruction execute

4. Briefly explain about multiplier accumulator.

The way to implement the correlation and convolution is array multiplication Method.
For getting down these operations we need the help of adders and multipliers. The combination of these accumulator and multiplier is called as multiplier accumulator.

5. What are the types of MAC is available?

There are two types MAC'S available

1. Dedicated & integrated
2. Separate multiplier and integrated unit

6. What is meant by pipeline technique?

The pipeline technique is used to allow overall instruction executions to overlap. That is where all four phases operate in parallel. By adapting this technique, execution speed is increased.

7. What are four phases available in pipeline technique?

The four phases are
(i) Fetch
(ii) Decode
(iii)Read
(iv)Execution

8. **In a non-pipeline machine, the instruction fetch, decode and execute take 30 ns, 45 ns and 25 ns respectively. Determine the increase in throughput if the instruction were pipelined**.

Assume a 5ns pipeline overhead in each stage and ignore other delays.
The average instruction time is = 30 ns+45 ns + 25 ns = 100 ns
Each instruction has been completed in three cycles = 45 ns * 3 = 135ns
Throughput of the machine =
The average instruction time/Number of M/C per instruction
= 100/135 = 0.7407
But in the case of pipeline machine, the clock speed is determined by the speed of the slowest stage plus overheads.
In our case is = 45 ns + 5 ns =50 ns
The respective throughput is = 100/50 = 2.00
The amount of speed up the operation is = 135/50 = 2.7 times

9. **Assume a memory access time of 150 ns, multiplication time of 100 ns, addition time of 100 ns and overhead of 10 ns at each pipe stage. Determine the throughput of MAC**

After getting successive addition and multiplications
The total time delay is 150 + 100 + 100 + 5 = 355 ns
System throughput is = 1/355 ns.

10.Write down the name of the addressing modes.

Direct addressing.
Indirect addressing.
Bit-reversed addressing.

Immediate addressing.

i. Short immediate addressing.
ii. Long immediate addressing.

Circular addressing.

11.What are the instructions used for block transfer in C5X Processors?

The BLDD, BLDP and BLPD instructions use the BMAR to point at the source or destination space of a block move. The MADD and MADS also use the BMAR to address an operand in program memory for a multiply accumulator operation

12.**Briefly explain about the dedicated register addressing modes.**

The dedicated-registered addressing mode operates like the long immediate addressing modes, except that the address comes from one of two special-purpose memory-mapped registers in the CPU: the block move address register (BMAR) and the dynamic bit manipulation register (DBMR).

The advantage of this addressing mode is that the address of the block of memory to be acted upon can be changed during execution of the program.

13. Briefly explain about bit-reversed addressing mode?

In the bit-reversed addressing mode, INDX specifies one-half the size of the FFT. The value contained in the current AR must be equal to 2n-1, where n is an integer, and the FFT size is 2n. An auxiliary register points to the physical location of a data value. When we add INDX t the current AR using bit reversed addressing, addresses are generated in a bit-reversed fashion. Assume that the auxiliary registers are eight bits long, that AR2 represents the base address of the data in memory (0110 00002), and that INDX contains the value 0000 10002.

14. Briefly explain about circular addressing mode.

Many algorithms such as convolution, correlation, and finite impulse response (FIR) filters can use circular buffers in memory to implement a sliding window; which contains the most recent data to be processed. The 'C5x supports two concurrent circular buffer operating via the ARs. The following five memory-mapped registers control the circular buffer operation.

1. CBSR1- Circular buffer 1 start register.
2. CBSR2- Circular buffer 2 start Register,
3. CBER1- Circular buffer 1 end register
4. CBER2- Circular buffer 2 end register
5. CBCR - Circular buffer control register.

15. Write the name of various part of C5X hardware.

1. Central arithmetic logic unit (CALU)
2. Parallel logic unit (PLU)
3. Auxiliary register arithmetic unit (ARAU)
4. Memory-mapped registers.
5. Program controller.

16. **Write short notes about arithmetic logic unit and accumulator**.

The 32-bit general-purpose ALU and ACC implement a wide range of arithmetic and logical functions, the majority of which execute in a single clock cycle. Once an operation is performed in the ALU, the result is transferred to the ACC, where additional operations, such as shifting, can occur. Data that is input to the ALU can be scaled by the prescaler.
The following steps occur in the implementation of a typical ALU instruction:
1. Data is fetched from memory on the data bus,
2. Data is passed through the prescaler and the ALU, where the arithmetic is performed, and
3. The result is moved into the ACC.

The ALU operates on 16-bit words taken from data memory or derived from immediate instructions. In addition to the usual arithmetic instructions, the ALU can perform Boolean operations, thereby facilitating the bit manipulation ability required of high-speed controller. One input to the ALU is always supplied by the ACC. The other input can be transferred from the PREG of the multiplier, the ACCB, or the output of the prescaler. After the ALU has performed the arithmetic or logical operation, the result is stored in the ACC.

17. Write short notes about parallel logic unit.

The parallel logic unit (PLU) can directly set, clear, test, or toggle multiple bits in control/status register pr any data memory location. The PLU provides a direct logic operation path to data memory values without affecting the contents of the ACC or the PREG.

18. What is meant by auxiliary register file?

The auxiliary register file contains eight memory-mapped auxiliary registers (AR0-AR7), which can be used for indirect addressing of the data memory or for temporary data storage. Indirect auxiliary register addressing allows placement of the data memory address of an instruction operand into one of the AR. The ARs are pointed to by a 3-bit auxiliary register pointer (ARP) that is loaded with a value from 0-7, designating AR0-AR7, respectively.

19. Write short notes about circular registers in C5X.

The ‘C5x devices support two concurrent circular buffers operating in conjunction with user-specified auxiliary register. Two 16-bit circular buffer start registers (CBSR1 and CBSR2) indicate the address where the circular buffer starts. Two 16-bit circular buffer end registers (CBER1 and CBER2) indicate the address where the circular buffer ends. The 16-bit circular buffer control register (CBCR) controls the operation of these circular buffers and identifies the auxiliary registers to be used.