SYED AMMAL ENGINEERING COLLEGE

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Department of EEE

Two Mark questions with Answers

Course Code and Name : EE 6403 Discrete Time systems and Signal Processing
Year and Semester : II year IV Semester
Prepared by : P.Nedumal Pugazhenthi, Associate professor/ EEE

UNIT-I - INTRODUCTION

1. **Define Signal.**
   A Signal is defined as any physical quantity that varies with time, space or any other independent variables.
   A signal can also be defined as a physical quantity which can be represented as a function of one or more variables which carries information.
2. **Define 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.
   - Process Industries- signal conditioning circuits in sensors and transducers
5. **Write the classifications of DT Signals.**
   - Deterministic & Random
   - Energy & Power signals
   - Periodic & Aperiodic signals
   - Even & Odd signals.
   - Causal & Non causal 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 objective 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 & Unstable system
- Static & Dynamic systems
- Time variant & Time invariant systems
- Recursive & Non Recursive systems
- FIR & IIR 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 and non causal signals.

The signal x[n] is defined only for $n \geq 0$ are termed as causal signal
The signal[n] is also defined for $n<0$ are termed as non causal signal

12. Define odd and even signal

Signal which satisfies the condition $x[-n]= x[n]$ is termed as even signal
Signal which satisfies the condition $x[-n]=-x[n]$ is called odd signal

13. Define deterministic and random signal

Signal which can be expressed mathematically can be called as deterministic signal and vice versa.

14. Define periodic and Aperiodic signals.

Signal which repeats its values or sequence after a particular time interval is said to be periodic.
Signal whose frequency is a multiple of π is also termed as periodic signal

15.Define causal and non causal systems

If the output of the system depends on the present and past but not on the future input is called causal system

If the output of the system depends on the future value of the input also it is termed as non causal system.

16.Define stable and unstable system

If the system responds with a bounded output for the bounded input the system is stable

$$\sum_{n=-\infty}^{\infty} h[n] < \infty$$

17.Define time variant and time invariant system

If the system satisfies the following condition it is called time invariant system

y[n,k]=y[n-k]

18.Define recursive and non recursive system

If the system output depends on present and past output and input values then it is called recursive system

y[n]=F[y(n-1),y(n-2)…y(n-m),x(n),x(n-1),…x(n-m)]

If the system output depends on present and past input but not on past output is called as non recursive system

y[n]=F[x(n),x(n-1),…x(n-m)]

19.Define IIR & FIR systems

In FIR system the impulse response consists of finite number of samples

$$y[n] = \sum_{m=0}^{N-1} h(m)x(n-m)$$

Where **h(n)=0 for n<0 and n≥N**

In IIR system the impulse response has infinite number of samples

$$y[n] = \sum_{m=0}^{\infty} h(m)x(n-m)$$

20.Why linear convolution is important in DSP?

The response of an LTI discrete time system for any input x[n] is given by linear convolution of the input x[n] and the impulse response of system which means that if the impulse response of the system is known then the response of the system for any input can be determined by convolution technique

21.What is zero padding?

Appending zeros to a sequence in order to increase the length of the sequence is called zero padding. This performed in circular convolution when the given two sequences are of different length.

22.List the difference between Linear and Circular Convolution

SLNO	Linear Convolution	Circular Convolution
1	The length of the input sequences can be different	The length of the input sequences should be same
2	Zero padding not required	Zero padding required if the sequences are of different length
3	The length of the output sequence can be calculated as Length of the first sequence- length of second sequence + 1 L1-L2+1	The length of the output sequence would have the same length of the input sequences

23.What are the types of convolutions?

- Linear convolution
- Circular convolution
- Sectioned convolution

24.Define sampling theorem

Sampling theorem is defined as " A band limited continuous time signal with highest frequency Fm hertz can be uniquely recovered from its samples provided that the sampling rate Fs is greater than or equal to 2Fm samples per second

Fs≥2Fm

Where Fm- maximum frequency of the signal to be sampled

Fs – Sampling frequency

25.Define Aliasing

The phenomenon of high frequency component getting the identity of low frequency component during sampling is called aliasing

26. What is sampling?

The process of converting a continuous valued continuous time signal into continuous valued discrete time signal is called sampling. It is done using sample and hold circuit.

27. What is quantization?

The process of converting continuous valued discrete time signal into discrete valued discrete time signal is called quantization. This is performed either by truncation or rounding off. The difference between quantized and unquantized signals is termed as quantization error.

28. What are the standard discrete time signals?

- Unit Impulse signal
- Unit step signal
- Ramp signal
- Exponential signal
- Sinusoidal signal
- Complex exponential signal

29. Define impulse and unit step signal.

<u>Impulse signal $\boldsymbol{\delta}$(n):</u>

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

$$\boldsymbol{\delta}[\boldsymbol{n}] = \begin{cases} \mathbf{1}; \boldsymbol{n} = \mathbf{0} \\ \mathbf{0}; \boldsymbol{n} \neq \mathbf{0} \end{cases}$$

<u>Unit step signal u(n):</u>

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

$$\boldsymbol{u}[\boldsymbol{n}] = \begin{cases} \mathbf{1}; \boldsymbol{n} \geq \mathbf{0} \\ \mathbf{0}; \boldsymbol{n} < 0 \end{cases}$$

30. What are the mathematical operations performed over discrete time signals?

- Scaling (Amplitude, Time)
- Folding
- Shifting
- Addition
- Multiplication

UNIT-II DISCRETE TIME SYSTEM ANALYSIS

1. **Define Z transform.**
The Z transform of a discrete time signal x(n) is denoted by X(z) and is given by

$$X[z] = \sum_{n=-\infty}^{\infty} x[n]z^{-n}$$

2. **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.
3. **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.
4. **List any four properties of Z-Transform.**
Linearity
Time Shifting
Frequency shift or Frequency translation
Time reversal
5. **What are the different methods of evaluating inverse z-transform?**
    - Partial fraction expansion
    - Power series expansion
    - Contour integration (Residue method)
6. **What are the properties of convolution?**
    - Commutative property x(n) * h(n) = h(n) * x(n)
    - Associative property [x(n) * h1(n)]*h2(n) = x(n)*[h1(n) * h2(n)]
    - Distributive property x(n) *[h1(n)+h2(n)] = [x(n)*h1(n)]+[x(n) * h2(n)]
7. **Define DTFT**.
Let us consider the discrete time signal x(n).Its DTFT is denoted as $X(e^{j\omega})$.
It is given as

$$X\left(e^{j\omega}\right) = \sum_{n=-\infty}^{\infty} x[n]e^{j\omega n}$$

8. **State the condition for existence of DTFT?**
The conditions is if x(n) is absolutely summable then

$$\sum_{n=-\infty}^{\infty} |x[n]| < \infty$$

9. **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

10. What is the DTFT of unit sample?
The DTFT of unit sample is 1 for all values of w.

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

12. Define step and impulse response of a discrete time system.
For a discrete time system the output response obtained for a step input it is termed as step response and for impulse input is impulse response.

13. Define convolution sum or discrete convolution or linear convolution.
It is the convolution of discrete time signals. It is used to obtain the output response of a discrete time system if the input signal x[n] and impulse response h[n] is known. It is denoted as

$$y[n] = \sum_{k=-\infty}^{\infty} x[k]h[n-k] \quad or \quad \sum_{k=-\infty}^{\infty} h[k]x[n-k]$$

14. State parseval's theorem.

$$E = \sum_{n=-\infty}^{\infty} |x[n]|^2 = \frac{1}{2\pi}\int_{-\pi}^{\pi} \left|X(e^{j\omega})\right|^2 d\omega$$

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

$$X(z) = 1 + 2z^{-1} + 3z^{-2} + 4z^{-3}$$

16. State frequency response of discrete time system

Consider the input signal to the discrete time system which is assumed to be a complex exponential signal $x[n] = e^{j\omega n}$

Then the output of the system is

$$y[n] = \sum_{k=-\infty}^{\infty} h[k] e^{j\omega(n-k)}$$

17.

$$= e^{j\omega n} \underbrace{\left[\sum_{k=-\infty}^{\infty} h[k] e^{-j\omega k} \right]}_{\text{Frequency response}}$$

Frequency response is denoted as

$$H(e^{j\omega}) = |H(e^{j\omega})| \angle H(e^{j\omega})$$

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 $\omega = 0 \; to \; 2\pi$
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. $X(k) = \sum_{n=0}^{N-1} x[n] e^{\frac{-j2\pi nk}{N}} \qquad k = 0,1, \ldots N-1$

4. **What is DFT of unit impulse δ(n)?**
   The DFT of unit impulse δ(n) is unity.
5. **List the properties of DFT.**
   Linearity, Periodicity, Circular symmetry, symmetry, Time shift, Frequency shift, complex conjugate, convolution, correlation and Parseval‘s theorem.
6. **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.
7. **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)
8. **What is the disadvantage of direct computation of DFT?**
   For the computation of N-point DFT, N^2 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.
9. **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.
   Radix-2 FFT algorithms. - Radix-2 Decimation in Time (DIT) algorithm. - Radix-2 Decimation in Frequency (DIF) algorithm.
   Radix-4 FFT algorithm.
10. **What is the computational complexity using FFT algorithm?**
    Complex multiplications = $N/2 \log_2 N$
    Complex additions = $N \log_2 N$

11. 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.

12. 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.

13. 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 32/2log232=16*5=80.

14. 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.

15. 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=2M, where M is an integer, Then this algorithm is known as radix-s FFT algorithm.

16. 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 $x_o(n)$, 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.

17. What are the differences and similarities between DIF and DIT algorithms?

DIT	DIF
The input is bit reversal while the output is in natural order	The input is in natural order while the output is bit reversed.
Complex multiplication is done first and then addition or subtraction is performed	complex multiplication is done after addition and subtraction
The structure of computation is $a \pm bW_N^K$	The structure of computation is $(a \pm b)W_N^K$

18. What are the applications of FFT algorithms?

- Linear filtering
- Correlation
- Spectrum analysis

19. 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.

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?

ω = 2 tan (w/2) T

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
Operates on digital samples of the signal.	Operates on analog signals
It is governed by linear difference equation.	It is governed by linear differential equation
It consists of adders, multipliers and delays implemented in digital logic	It consists of electrical components like resistors, capacitors and inductors

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. **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.
17. **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.

18. Why an impulse invariant transformation is not considered to be one-to one?

In impulse invariant transformation any strip of width 2π/T in the s-plane for values of s-plane in the range $(2k-1)/T < \Omega < (2k-1)\pi/T$ 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 zplane, 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.

19. 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

20. **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.

21. Write the properties of Chebyshev type –1 filters.

The magnitude response is equiripple in the passband and monotonic in the stopband.

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.

22. **Compare the Butterworth and Chebyshev Type-1 filters**.

Butterworth	Chebyshev Type - 1
All pole design	All pole design
The poles lie on a circle in s-plane	The poles lie on a ellipse in s-plane.
The magnitude response is maximally flat at the origin and monotonically decreasing function of ωc.	The magnitude response is equiripple in pass band and monotonically decreasing in the stop band
The normalized magnitude response has a value of $1 / \sqrt{2}$ at the cutoff frequency ωc.	The normalized magnitude response has a value of $1 / \sqrt{(1+\epsilon 2)}$ at the cutoff frequency ωc.
Only few parameters has to be calculated to determine the transfer function.	A large number of parameters has to be calculated to determine the transfer function

23. 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.

24. 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.

25. 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.

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

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

28. 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,

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

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

30. 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.

31. How phase distortion and delay distortions are introduced?

32. 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.

33. 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.

34. 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 non recursively are always stable.

The roundoff noise can be made small in nonrecursive realization of FIR filters.

35. 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.

36. 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.

37. 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)

38. 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)

39. 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.

40. **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.

41. 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 pass band and stop band. This effect is known as Gibb‘s phenomenon (or Gibb‘s Oscillation).

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

43. 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.

44. 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.

45. 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.

46. 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 main lobe for a given peak side lobe amplitude.

47. List the desirable features of Kaiser Window spectrum.

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

UNIT V - DIGITAL SIGNAL PROCESSOR

1. What are the classifications of Digital Signal Processors?

They are classified into

- General purpose digital signal processor
- Special Purpose digital signal processor

2. What are the factors that influence selection of DSPs?

- Architectural features
- Execution speed
- Type of arithmetic
- Word Length

3. 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.

4. **List the types of special purpose DSP processors.**

There are two types of special purpose DSP processor

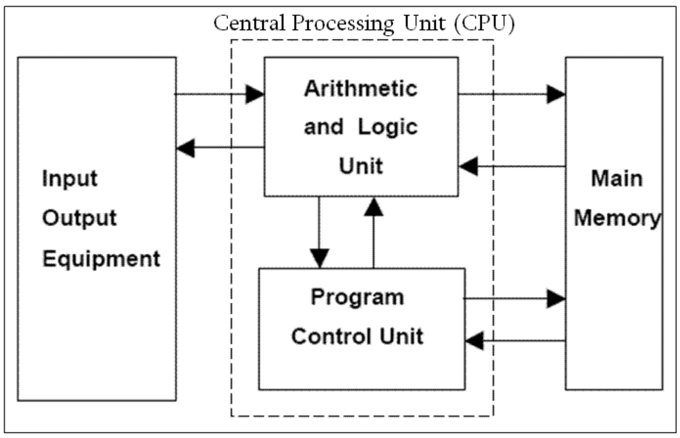
- Hardware designed for efficient execution of specific DSP algorithms such as digital filter, FFT.
- Hardware designed for specific applications, for example telecommunication, digital audio.

5. **Briefly explain about Harvard architecture.**

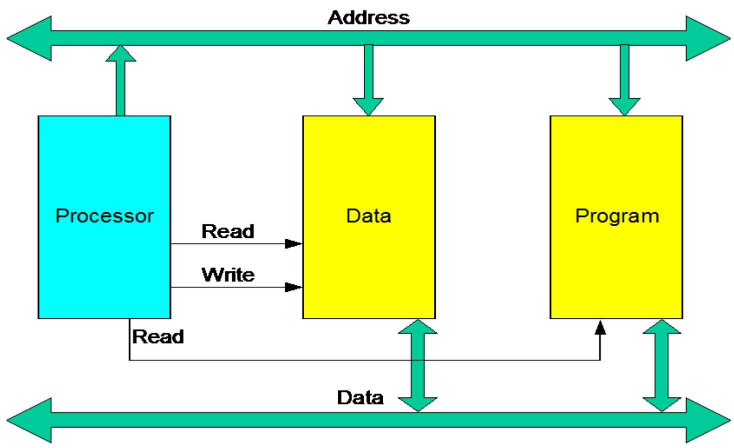
The principal feature of Harvard architecture is that the program and the data memories lie in two separate spaces, permitting full overlap of instruction fetch and execution. Typically these types of instructions would involve their distinct type.

- o Instruction fetch
- o Instruction decode
- o Instruction execute

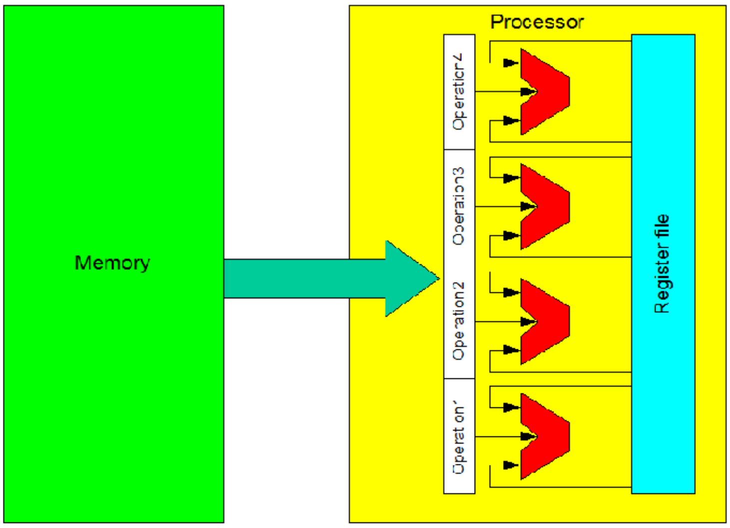
6. **Draw the Von Newmann Architecture.**



7. **Draw the Harvard architecture.**



8. **Draw the VLIW architecture**



9. **What are the types of MAC is available?**

There are two types MAC‘S available

- Dedicated & integrated
- Separate multiplier and integrated unit

10. What are four phases available in pipeline technique?

The four phases are

- Fetch
- Decode
- Read
- Execution

11. **What is pipelining depth?**

The number of pipelining stages is referred to as the pipelining depth

12. **Write down the name of the addressing modes**.

- Direct addressing
- Indirect addressing
- Bit-reversed addressing
- Immediate addressing.
- Circular addressing.

13. 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

14. 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.

15. 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.

16. **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.

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

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

- Central arithmetic logic unit (CALU)
- Parallel logic unit (PLU)
- Auxiliary register arithmetic unit (ARAU)
- Memory-mapped registers.
- Program controller.

18. 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:

- Data is fetched from memory on the data bus,
- Data is passed through the prescaler and the ALU, where the arithmetic is performed, and
- 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.

19. Write short notes about parallel logic unit.

The parallel logic unit (PLU) can directly set, clear, tests, 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.

20. 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.

21. 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.

22. What are the factors that influence selection of DSPs?

- Architectural features
- Execution speed
- Type of arithmetic
- Word length

23. What are the applications of PDSPs?

Digital cell phones, automated inspection, voicemail, motor control, video conferencing, noise cancellation, medical imaging, speech synthesis, satellite communication etc.

24. Give some examples for fixed point DSPs.

TM32OC50, TMS320C54, TMS320C55, ADSP-219x, ADSP-219xx..

25. Give some example for floating point DSPs?

TMS320C3x, TMS320C67x, ADSP-21xxx

26. What is pipelining?

Pipelining a processor means breaking down its instruction into a series of discrete pipeline stages which can be completed in sequence by specialized hardware.

27. What are the advantages of VLIW architecture?

Advantages of VLIW architecture

- Increased performance
- Better compiler targets
- Potentially easier to program
- Potentially scalable

- Can add more execution units; allow more instructions to be packed into the VLIW instruction.

28. What are the disadvantages of VLIW architecture?

Disadvantages of VLIW architecture

- New kind of programmer/compiler complexity
- Program must keep track of instruction scheduling
- Increased memory use
- High power consumption

29. What is the pipeline depth of TMS320C50 and TMS320C54x?

- TMS320C50 – 4
- TMS320C54x – 6

30. What are the different buses of TMS320C5x?

The C5x architecture has four buses

- Program bus (PB)
- Program address bus (PAB)
- Data read bus (DB)
- Data read address bus (DAB)

31. Give the functions of program bus?

The program bus carries the instruction code and immediate operands from program memory to the CPU.

32. Give the functions of program address bus?

The program address bus provides address to program memory space for both read and write.

33. Give the functions of data read bus?

Data read bus interconnects various elements of the CPU to data memory space.

34. Give the functions of data read address bus?

The data read address bus provides the address to access the data memory space.

35. What are the different stages in pipelining?

- The fetch phase
- The decode phase
- Memory read phase
- The execute phase

36. List the various registers used with ARAU.

- Eight auxiliary registers (AR0 – AR7)
- Auxiliary register pointer (ARP)
- Unsigned 16-bit ALU

37. What are the elements that the control processing unit of ‘C5x consists of ?

The central processing unit consists of the following elements:

- Central arithmetic logic unit (CALU)
- Parallel logic unit (PLU)
- Auxiliary register arithmetic unit (ARAU)
- Memory mapped registers

- Program controller

38. What is the function of parallel logic unit?

The parallel logic unit is a second logic unit that executes logic operations on data without affecting the contents of accumulator.

39. List the on chip peripherals in 'C5x.

The on-chip peripherals interfaces connected to the C5x CPU include

- Clock generator
- Hardware timer
- Software programmable wait state generators
- General purpose I/O pins
- Parallel I/O ports
- Serial port interface
- Buffered serial port
- Time-division multiplexed (TDM) serial port
- Host port interface
- User unmask able interrupts

40. What are the arithmetic instructions of 'C5x?

ADD, ADDB, ADDC, SUB, SUBB, MPY, MPYU

41. What are the shift instructions?

ROR, ROL, ROLB, RORB, BSAR.

42. What are the general purpose I/O pins?

- Branch control input (BIO)
- External flag (XF)

43. What are the logical instructions of 'C5x?

AND, ANDB, OR, ORB, XOR, XORB

44. What are load/store instructions?

LACB, LACC, LACL, LAMM, LAR, SACB, SACH, SACL, SAR, SAMM.

45. Mention the addressing modes available in TMS320C5X processor?

- Direct addressing mode
- Indirect addressing mode
- Circular addressing mode
- Immediate addressing
- Register addressing
- Memory mapped register addressing

46. What is function of NOP instruction?

NOP- No operation

Perform no operation.

47. What is function of ZAC instruction?

ZAC – Zero accumulator

Clear the contents of accumulator to zero.

48. Give the function of BIT instruction.

BIT – Test bit Copy the specified bit of the data memory value to the TC bit in ST1.

49. What is use of ADD instruction?

ADD – Add to accumulator with shift.

Add the content of addressed data memory location or an immediate value of accumulator, if a shift is specified, left-shift the data before the add. During shifting, low order bits are Zero-filled, and high-order bits are sign extended if SXM=1.

50. Give the advantages of DSPs?

Architectural features, Execution speed, Type of arithmetic, Word length.