19ECB212 - DIGITAL SIGNAL PROCESSING

TWO MARKS QUESTIONS AND ANSWERS

UNIT – I DISCRETE FOURIER TRANSFORM

1. Define DFT of a discrete time sequence.

The DFT is used to convert a finite discrete time sequence x(n) to an N-point frequency domain sequence denoted by X(k). The N-point DFT of a finite duration sequence x(n) of length L, where L ≤ N is defined as.

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

2. Define IDFT.

The IDFT is used to convert the N-point frequency domain sequence X(k) to an N-point time domain sequence. The IDFT of the sequence X(k) of length n is defined as

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

3. What is the relation between DTFT and DFT.

Let x(n) be a discrete time sequence. Now DTFT[x(n)] = X(ω) or FT[x(n)] = X(ω) and DFT[x(n)] = X(k). The X(ω) is a periodic continuous function of ω and X(k) is an N – point periodic sequence. The N –point sequence x(k) is actually N samples of X(ω) which can be obtained by sampling one period of X(ω) at N equal intervals.

4. What is the draw back in Fourier Transform and how it is overcome.

The drawback in Fourier Transform is that it is a continuous function of ω and so it cannot be processed by digital system. This drawback is overcome by using Discrete Fourier transform. The DFT converts the continuous function of ω to a discrete function of ω.

5. **Write two applications of DFT**.

   (a)The DFT is used for spectral analysis of signals using a digital computer.

   (b)The DFT is used to perform filtering operations on signals using digital computer.

6. **When an N- point periodic sequences is said to be even or odd sequence.**

   An N – point periodic sequence is called even if it satisfies the condition.

   $X(n-N) = x(n)$; for $0 \le n \le (N-1)$

   An N – point periodic sequence is called odd if it satisfies the condition.

   $X(n-N) = -x(n)$; for $0 \le n \le (N-1)$

7. **List any four properties of DFT.**

Let DFT{x(n)} = X(k), DFT{x1(n)} = X1(k) and DFT{x2(n)} = X2(k)

a. Periodicity: $X(k+N) = X(k)$; for all k
b. Linearity: DFT{a1 x1(n)+ a2 x2(n)} = a1 X1(k)+ a2 X2(k); where a1 and a2 are constants.
c. DFT of time revised sequence: DFT{x(N-n)} =X(N-k)
d. Circular Convolution: DFT {x1(n) ʘ x2(n)} = X1(k) X2(k)

8. **Why linear convolution is important in DSP.**

   The response or output of LTI discrete time system for any input x(n) is given by linear convolution of the input x(n) and the impulse response h(n) of the system. This means that if the impulse response of a system is known, then the response for any input can be determined by convolution operation.

9. **Write the properties of Linear Convolution**.

   a. Commutative Property: x(n) * h(n) = h(n) * x(n)
   b. Associative Property: [x(n) * h1(n)] * h2(n) = x(n) * [h1(n) * h2(n)]
   c. Distributive Property: x(n) * [h1(n) + h2(n)] = [x(n) * h1(n)] + [x(n) * h2(n)]

10. What is Zero Padding? Why it is needed.

Appending Zeros to a sequence in order to increase the size or length of the sequence is called Zero Padding. In circular convolution, when the two input sequences are different size, then they are converted to equal size by zero padding.

11. Compare the Overlap Add and Overlap Save method of sectioned convolutions.

Overlap Add method	Overlap Save method
1) Linear convolution of each section of longer sequence with smaller sequence is performed.	Circular convolution of each section of longer sequence with smaller sequence is performed. (After converting them to the size of output sequence)
2) Zero Padding is not required.	Zero Padding is required to convert the input sequences to the size of the output sequence.
3) Overlapping of samples of input sections are not required.	The N2-1 samples of an input section of longer sequence is overlapped with next input section.
4) The overlapped samples in the output of sectioned convolutions are added to get the overall output.	Depending on the method of overlapping the input samples, either the last N2-1 samples or the first N2-1 samples of the output sequence of each sectioned convolutions are discarded.

12. In what way Zero padding is implemented in overlap save method.

In overlap save method, the Zero padding is employed to convert the smaller input sequence to the size of the output sequence of each sectioned convolution. The zero padding is also employed to convert either the last or first section of the longer input sequence to the size of the output sequence of each sectioned convolution. This depends on the method of overlapping input samples.

13. What is FFT?

The FFT is a method for computing the DFT with reduced number of calculations. The computational efficiency is achieved by divide and conquer approach. This is based on the decomposition of an N – point DFT into successively smaller DFT's.

14. What is radix – 2 FFT?

The radix – 2 FFT is an efficient algorithm for computing N – point sequence. In Radix – 2 FFT the N – point sequence is decimated in to 2 – point sequences 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. What is DIT radix – 2 FFT?

The Decimation in Time (DIT) radix – 2 FFT is an efficient algorithm for computing DFT. In DIT radix – 2 FFT, the time domain N – point sequence is decimated into 2 – point sequences. The results of 2 – point DFT's are used to compute 4 – point DFT. Two numbers of 2 – point DFT's are combined to get 4 – point DFT. The result of 4 – point DFT's are used to compute 8 – point DFT. Two numbers of 4– point DFT's are combined to get a 8 – point DFT.

This process is continued until we get N point DFT.

16. What is phase factor or twiddle factor.

The complex number WN is called phase factor or twiddle factor. The W_N represent a complex number e-j2π/n. It also represent an Nth root of unity.

17. What is DIF radix – 2 FFT.

The Decimation in Frequency (DIF) radix – 2 FFT is an efficient algorithm for computing DFT. In this algorithm the N – point time domain sequence is converted in to two numbers of N/2 point sequences. Then each N/2 point sequence is converted to two numbers of N/4 point sequences. This process is continued until we get N/2 numbers of 2 – point sequences. Now the 2 – point DFT's of N/2 numbers of 2 – point sequences will give n samples, which is the N – point DFT of the time domain sequence. Here the equations forming N/2 point sequences, N/4 point sequences etc.; are obtained by decimation of frequency domain sequences.

18. Compare the DIT and DIF radix – 2 FFT.

DIT radix – 2 FFT	**DIF radix – 2 FFT**
1. The time domain sequence is decimated. 2. When the input is in bit reversed order, the output will be in normal order and vice versa.	The frequency domain sequence is decimated. When the input is in bit normal order, the output will be in bit reversedorder and vice versa.

In each stage of computations, the phase factors are multiplied before add and subtract operations.	In each stage of computations, the phase factors are multiplied after add and subtract operations.
4. The value of N should be expressed such that N = 2m and this algorithm consists of m stages of computations.	The value of N should be expressed such that N = 2m and this algorithm consists of m stages of computations.
5. Total number of arithmetic operations are Nlog2N complex additions and (N/2)log2N complex multiplications.	Total number of arithmetic operations are Nlog2N complex additions and (N/2)log2N complex multiplications.

19. Distinguish between DFT and DTFT.

DFT	DTFT
. Obtained by performing sampling operation in both the time and frequency domains.	Sampling is performed only in time domain.
. Discrete frequency spectrum.	Continuous function of ω.

20. Distinguish between linear convolution and circular convolution of two sequences.

Linear Convolution	Circular Convolution
1. If x(n) is a sequence of L number of samples and h(Linear n) with m number of samples, after convolution y(n) will contain N = L + M – 1 samples	If x(n) is a sequence of L number of samples and h(n) with m number of samples, after convolution y(n) will contain N = Max(L,M) samples
2.Linear convolution can be used to find the response of a linear filter.	Circular convolution can be used to find the response of a linear filter
3. Zero padding is not necessary to find the response of a linear filter.	Zero padding is necessary to find the response of a linear filter.

21. Find DFT of the sequence x(n)={1,1,0,0}

$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi kn/N}$, k=0,1,……..N-1

$= \sum_{n=0}^{3} x(n) e^{-j2\pi kn/2}$, $\quad k = 0,1, \dots \dots 3$

X(0) $= \sum_{n=0}^{3} x(n) = \{ 1 + 1 + 0 + 1\} = 2$

X(1) $= \sum_{n=0}^{3} x(n) e^{-j\pi n/2} = \{1 - j + 0 + 0\} = 1 - j$

X(2) $= \sum_{n=0}^{3} x(n) e^{-j\pi n} = \{1 - 1 + 0 + 0\} = 0$

X(3) $= \sum_{n=0}^{3} x(n) e^{-j3\pi n/2} = \{1 + j + 0 + 0\} = 1 + j$

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

22. Define Circular Convolution?

Let $x_1(n)$ and $x_2(n)$ are finite duration sequences both of length N with DFTs $X_1(k)$ and $X_2(k)$.

If $X_3(k) = X_1(k)\ X_2(k)$, then the sequence $x_3(n)$ can be obtained by circular convolution defined as

$$\mathbf{x_3\ (n)} = \sum_{m=0}^{N-1} x_1(m) x_2((n-m))_N$$

23. How the circular convolution is obtained?

Given two sequences $x_1(n)$ and $x_2(n)$, the circular convolution of these two

sequences $x_3(n) = x_1(n)$ Ⓝ $x_2(n)$ can be obtained by using the following steps.

1. Graph N samples of $x_1(n)$, as equally spaced points around an outer circle in counter clockwise direction.
2. Start at the sample point as $x_1(n)$ graph N samples of $x_2(n)$ as equally spaced points around an inner circle in clockwise direction.
3. Multiply corresponding samples on the two circles and sum the products to produce output.
4. Rotate the inner circle one sample at a time in clockwise direction and go to step 3 to obtain the next value of output.
5. Repeat step.no.4 until the inner circle first sample lines up with the first sample of the exterior circle again.

UNIT – II INFINITE IMPULSE RESPONSE DIGITAL FILTERS

1. Define an IIR filter.

The filters designed by considering all the infinite samples of impulse response are called IIR filters. The impulse response is obtained by taking inverse Fourier transform of ideal frequency response.

2. Compare IIR and FIR filter

IIR	FIR
All the infinite samples of impulse response are considered	Only N samples of Impulse response is considered
The Impulse response cannot be directly converted to digital transfer function	The impulse response can be directly converted to digital transfer function
The design involves design of analog filter and then transferring the analog filter into digital filter is done	The digital filter can be directly designed to achieve the desired specifications.

3. What are the advantages and disadvantages of digital filters. Advantages of digital filters

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

Disadvantages of digital filters

- 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.

4. What are the requirements for an analog filter to be stable and causal.

The analog filter transfer function Ha(s) should be a rational function of s and the coefficients of s should be real.

The poles should lie on the left half of s – plane.

The number of zeroes should be less than or equal to number of poles.

5. What are the requirements for an digital filter to be stable and causal .

The analog filter transfer function H (z) should be a rational function of z and the coefficients of z should be real.

The poles should lie inside the unit circle in z - plane.

The number of zeroes should be less than or equal to number of poles.

6. Compare the digital and analog filter.

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

7. What is impulse invariant transformation?

The transformation of analog filter to digital filter without modify the impulse response of the filter is called impulse invariant transformation. In this transformation the impulse response of the digital filter will be sampled version of the impulse response of the analog filter.

8. 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 the 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

S **= 2/T (1 – z-1/1 + z-1)**

9. What is frequency warping?

In bilinear transformation the relation between analog and digital frequencies is non-linear. When the s – plane is mapped into z – plane using bilinear transformation, this non-linear relationship introduces distortion in frequency axis, which is called frequency warping.

10. What are the advantages and disadvantages of bilinear transformation?

Advantages of bilinear transformation

- The bilinear transformation is one –to-one mapping.
- There is no aliasing and so the analog filter need not have a band limited frequency response.
- The effect of warping on amplitude response can be eliminated by prewarping the analog filter.
- It can be used to design digital filters with prescribed magnitude response with piecewise constant values.

Disadvantages of bilinear transformation

- The nonlinear relationship between analog and digital frequencies introduces frequency distortion which is called frequency warping.
- Using bilinear transformation, a linear phase analog filter cannot be transformed to linear phase digital filter.

11. What is prewarping. Why it is employed.

In IIR filter design using bilinear transformation, the conversion of the specified digital frequencies to analog frequencies is called prewarping.

It is necessary to eliminate the effect of warping on amplitude response.

12. Explain the technique of prewarping.

In IIR filter design using bilinear transformation, the conversion of the specified digital frequencies to analog frequencies is called prewarping.

Using the prewar frequencies, the analog filter transfer function is designed and then it is transformed to digital filter transfer function.

13. Compare the impulse invariant and bilinear transformations

	Impulse Invariant Transformation	Bilinear transformation
i.	It is many-to-one mapping	It is one-to-one mapping
ii.	The relation between analog and digital frequency is linear.	The relation between analog and digital frequency is non-linear.
iii.	To prevent the problem of aliasing the analog filters should be band limited.	Is no problem of aliasing and so the analog filters need not be band limited. Due to the effect of warping, the phase

iv. The magnitude and phase response of analog filter can be preserved by choosing low sampling time or high sampling frequency.	response of analog filters cannot be preserved. But the magnitude response can be preserved by prewarping.

14. What is Butterworth approximation.

In Butterworth approximation, the error function is selected such that the magnitude is maximally flat in the origin (i.e., at $\Omega = 0$) and monotonically decreasing with increasing Ω.

15. Write the properties of Butterworth filter.

1. The Butterworth filters are all pole designs.
2. At the cutoff frequency Ωc, the magnitude of normalized Butterworth filter is $1/\sqrt{2}$.
3. The filter order n, completely specifies the filter and as the value of N increases the magnitude response approaches the ideal response.

16. What is Chebyshev approximation.

In Chebyshev approximation, the approximation function is selected such that the error is minimized over a prescribed band of frequencies.

17. What is type – I Chebyshev approximation.

In type – I Chebyshev approximation, the error function is selected such that, the magnitude response is eqiuripple in the pass band and monotonic in the stop band.

18. What is type – II Chebyshev approximation.

In type – II Chebyshev approximation, the error function is selected such that, the magnitude response is monotonic is pass band and equiripple in stop band. The type– II magnitude response is called inverse Chebyshev response.

19.Write the properties of Chebyshev type – I filters.

- The magnitude response is equiripple in the pass band and monotonic in the stop band.
- The Chebyshev type – I filters are all pole designs.
- The normalized magnitude function has a value of 1/√1+€2 at the cutoff

frequency Ωc. The magnitude response approaches the ideal response as the value of N increases.

20. What are the different types of structures for realization of IIR systems. The different types of structures for realization of IIR system are.

i. Direct – form I structure,

ii. Direct – form II structure,

iii. Cascade form structure,

iv. Parallel form structure

21. Give the equation for the order of N and cutoff frequency Ωc of Butterworth filter.

The order of the filter

$$N = \frac{log\sqrt{\frac{10^{0.1\alpha_S - 1}}{10^{0.1\alpha_p - 1}}}}{log(\frac{\Omega s}{\Omega p})}$$

Where α_s = stop band attenuation at stop band frequency Ωs

α_p = pass band attenuation at stop band frequency Ωp

Cutoff frequency Ωc = $\frac{\Omega p}{(10^{0.1\alpha_p - 1})^{\frac{1}{2N}}}$

22.Give the expression for location of poles of normalized Butterworth filter

The poles of the Butterworth filter is given by

$$s_k = e^{j\varphi_k},\ k = 1, \ldots . N$$

$$\varphi_k = \frac{\pi}{2} + \frac{(2k-1)\pi}{2N}$$

N is order of the filter.

23. Distinguish between Butterworth Chebyshev (Type –I) filter.

1. The magnitude response of Butterworth filter decreases monotonically as the frequency Ω increases from 0 to ∞, whereas the magnitude response of the Chebyshev filter exhibits ripple in the pass band and monotonically decreasing in the stopband

2. The transition band is more in Butterworth filter compared to Chebyshev filter.

3.The poles of the Butterworth filter lie on a circle whereas the poles of the Chebyshev filter lie on an ellipse .

4. For the same specifications the number of poles in Butterworth filter are more when compared to Chebyshev

24. Mention any two procedures for digitizing

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

1. Impulse Invariant method.
2. Bilinear Transformation method.

25. What are the properties that are maintained same in the transfer of analog filter into a digital filter?

(a) The jΩ axis in the s-plane should map into the unit circle in the z-plane. Thus there is a direct relationship between the two frequency variables in the two domain.

(b) The left half of the s-plane should map into the inside of the unit circle in the z-plane .Thus a stable analog filter will be converted to a stable digital filter.

26. Physically realizable and stable IIR filters cannot have linear phase. Prove

Linear phase must have a system function that satisfies the condition

$$H(z) = \pm z^{-N} H(z^{-1})$$

Where z^{-N} represents a delay of N units of unit of time. But if this is ths case , for every pole inside the unit circle there must be a pole outside the unit circle.

Hence the filter would be unstable. Consequently a causal and stable IIR filter cannot have linear phase.

26. Give the bilinear transform equation between s-plane and z-plane .

$$\boldsymbol{s = \frac{2}{T}\left[\frac{1-z^{-1}]}{1+z^{-1}}\right]}$$

27. Distinguish between recursive realization and non recursive realization.

For recursive realization the present output y(n) is a function of past output and present inputs. This form corresponds to an infinite Impulse response (IIR) digital filter.

For non recursive realization the current output y(n) is a function of only past and present inputs. This form corresponds to an Finite Impulse Response (FIR) digital filter.

28. What is the main disadvantage of direct form realization?

The direct form realization is extremely sensitive to parameter quantization. When the order of the system N is large, a small change in a filter quantization due to parameter quantization, results in a large change in the location of the poles and zeros of the system.

29. What is the advantage of cascade form realization?

Quantization errors can be minimized if we realize an LTI system in cascade form.

30. Define Signal Flow graph.

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

UNIT – III FINITE IMPULSE RESPONSE DIGITAL FILTERS

1. What are FIR filters?

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

2. Write the steps involved in FIR filter design.

i. Choose the desired frequency response Hd(ω).
ii. Take inverse Fourier transform of Hd(ω) to get hd(n).
iii. Convert the infinite duration hd(n) to finite duration sequence h(n).
iv. Take Z - transform of h(n) to get the transfer function H(z) of the FIR filter.

3. What are the advantages of FIR filters.

- Linear phase FIR filters can be easily designed.
- Efficient realizations of FIR filter exist as both recursive and non-recursive structures.
- FIR filters realized non-recursively are always stable.
- The round off noise can be made small in non-recursive realization of FIR filters.

4. What are the dis-advantages in FIR filters.

i. The duration of impulse response should be large to realize sharp cut off filters.
ii. The non-integral delay can be lead to problems in some signal processing applications.

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

i. Symmetric impulse response when N is odd.
ii. Symmetric impulse response when N is even.
iii. Anti-Symmetric impulse response when N is odd.
iv. Anti-Symmetric impulse response when N is even.

6. **List the well known design techniques for linear phase FIR filter.**

   There are three well known methods of design techniques for linear phase FIR filters. They are,

   i. Fourier series method and Window method.

   ii. Frequency sampling method.

   iii. Optimal filter design methods.

7. **What is Gibb's phenomenon.**

   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.

8. **Write the procedures for designing FIR filter using Windows.**

- Choose the desired frequency response Hd(ω).
- Take inverse Fourier transform of Hd(ω) to get hd(n).
- Choose a window sequence w(n) and multiply hd(n) by w(n) to convert the infinite duration impulse response to finite duration impulse response h(n).
- The transfer function H(z) of the filter is obtained by taking Z – transform of h(n).

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

The desirable characteristics of the frequency response of window function. 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 ω tends to π.

i. Write the procedure for FIR filter design by frequency sampling Choose the desired frequency response Hd(ω).

ii. Take N – samples of Hd(ω) to generate the sequence $\hat{H}(k)$.

iii. Take inverse DFT of Hd(ω) to get the impulse response h(n).

iv. The transfer function H(z) of the filter is obtained by taking Z – transform of impulse response.

10. Write the characteristics of rectangular window.

- The main lobe width is equal to 4π/N.
- The maximum side lobe magnitude is – 13dB.
- The side lobe magnitude does not decreases significantly with increasing

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

- The width of the transition region is related to the width of the main lobe of window spectrum.
- Gibb's oscillations are noticed in the pass band and stop band.
- The attenuation in the stop band is constant and cannot be varied.

12. 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.

13. List the characteristics features of triangular window.

The main lobe width is equal to 8π/N.

The maximum side lobe magnitude is – 25dB.

The side lobe magnitude slightly decreases with increasing

14. List the features of hanning window spectrum.

The main lobe width is equal to 8π/N.

The maximum side lobe magnitude is – 31dB.

The side lobe magnitude decreases with increasing ϖ.

15. List the features of blackmann window spectrum.

The main lobe width is equal to 12π/N.

The maximum side lobe magnitude is – 58dB.

The side lobe magnitude decreases with increasing ϖ.

16. List the features of hamming window spectrum.

The main lobe width is equal to 8π/N.

The maximum side lobe magnitude is – 41dB.

The side lobe magnitude remains constant for increasing.

17. What are the factors that influence the choice of structure for realization f an LTI system?

The factors that influence the choice of realization structure are computational complexity, memory requirements, finite word length effects, parallel processing and pipelining computations.

18. What is the advantage in direct form – I structure when compared to direct form – II structure.

In direct form – II structure the number of delay element required is exactly half that direct form – I structure when the number of poles and zeros are equal. Hence it requires less amount of memory.

19. What are the difficulties in Cascade realization?

The difficulties in Cascade realization.

i. Decision of pairing poles and zeros.
ii. Deciding the order of cascading the first and second order sections.
iii. Scaling multipliers should be provided between individual sections to prevent the filter variables from becoming too large or too small.

20. What are the different types of filter based on impulse response?

Based on impulse response, 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.

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

The filters can be classified based on frequency response as (i) low pass filter (ii) high pass filter (iii) bandpass filter (iv) band reject filter.

22. What is the most general form of IIR filter?

The most general form of IIR filter can be written as

$$H(z) = \frac{\sum_{i=0}^{M} b_k z^{-k}}{1+\sum_{i=1}^{N} a_k z^{-k}}$$

23. What do you understand by linear phase response?

For a linear phase filter θ(ω) α ω, the linear phase filter does not alter the shape of the original signal. If the phase response of the filter is non linear then the output signal may be a distorted one. Linear phase response is preserved throughout pass band in order to preserve the shape of the signal within the pass band. The FIR filter will have linear phase when the impulse response of the filter is symmetric about the midpoint.

24. **Suppose the axis of symmetry of impulse response h(n) lies halfway between two samples, for what kind of applications this type of impulse response is used.**

In the axis of symmetry lies midway between two samples, such type of sequences can be used to design Hilbert transformer and differentiators.

24. For what kind of applications, the anti symmetrical impulse response can be used?

The anti symmetrical impulse response can be used to design Hilbert transformers and differentiators.

25. For what kind of application, the symmetrical impulse response can be used?

The impulse response, which is symmetric having odd number of samples can be used to design all types of filters , i.e ., low pass, high pass , band pass and band reject.

26. What condition on the FIR sequence h(n) are to be imposed in order that this filter can be called a linear phase filter?

The conditions are

(i) Symmetric condition h(n)=h(N-1-n)

(ii) Antisymmetric condition h(n)=-h(N-1-n).

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

If the impulse response is antisymmetrical , satisfying the condition h(n)= -h(N-1-n) the frequency response of FIR filter will have constant group delay and not the phase delay.

28. 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 .i.e

$$\sum_{n=-\infty}^{n=\infty} |h(n)| < \infty$$

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.

30. What are the disadvantages of Fourier series method?

In designing FIR filter using Fourier series method the infinite duration impulse response is truncated at $\omega = \pm \frac{(N-1)}{2}$. Direct truncation of the series will lead to fixed percentage overshoots and undershoots before and after an approximated discontinuity in the frequency response.

UNIT – IV FINITE WORD LENGTH EFFECTS

1. What do finite word length effects mean?

The effects due to finite precision representation of numbers in a digital system are called finite word length effects.

2. List some of the finite word length effects in digital filters.

1. Errors due to quantization of input data.

2. Errors due to quantization of filter co-efficient

3. Errors due to rounding the product in multiplications

4. Limit cycles due to product quantization and overflow in addition.

3. What are the different formats of fixed-point representation?

a. Sign magnitude format

b. One's Complement format

c. Two's Complement format.

In all the three formats, the positive number is same but they differ only in representing negative numbers.

4. Explain the floating-point representation of binary number.

The floating-point number will have a mantissa part. In a given word size the bits allotted for mantissa and exponent are fixed. The mantissa is used to represent a binary fraction number and the exponent is a positive or negative binary integer. The value of the exponent can be adjusted to move the position of binary point in mantissa. Hence this representation is called floating point.

5. What are the types of arithmetic used in digital computers?

The floating point arithmetic and two's complement arithmetic are the two types of arithmetic employed in digital systems.

6. What are the two types of quantization employed in digital system?

The two types of quantization in digital system are Truncation and Rounding.

7. What is truncation?

The truncation is the process of reducing the size of binary number by discarding all bits less significant than the least significant bit that is retained. In truncation of a binary number of b bits all the less significant bits beyond bth bit are discarded.

8. What is rounding?

Rounding is the process of reducing the size of a binary number to finite word size of b-bits such that, the rounded b-bit number is closest to the original unquantized number.

9. Explain the process of upward rounding?

In upward rounding of a number of b-bits, first the number is truncated to b-bits by retaining the most significant b-bits. If the bit next to the least significant bit that is retained is zero, then zero is added to the least significant bit of the truncated number. If the bit next to the least significant bit that is retained is one then one is added to the least significant bit of the truncated number.

10. What are the errors generated by A/D process?

The A/D process generates two types of errors. They are quantization error and saturation error. The quantization error is due to representation of the sampled signal by a fixed number of digital levels. The saturation errors occur when the analog signal exceeds the dynamic range of A/D converter.

11. What is quantization step size?

In digital systems, the numbers are represented in binary. With b-bit binary we can generate 2^b different binary codes. Any range of analog value to be represented in binary should be divided into 2^b levels with equal increment. The 2^b levels are called quantization levels and the increment in each level is called quantization step size. If R is the range of analog signal then, Quantization step size, $\mathbf{q = R/2^b}$

12. Why errors are created in A/D process?

In A/D process the analog signals are sampled and converted to binary. The sampled analog signal will have infinite precision. In binary representation of bbits we have different values with finite precision. The binary values are called quantization levels. Hence the samples of analog are quantized in order to fit into any one of the quantized levels. This quantization process introduces errors in the signal.

13. What is steady state output noise power due to input quantization?

The input signal to digital system can be considered as a sum of unquantized signal and error signal due to input quantization. The response of the system can be expressed as a summation of response due to unquantized input and error signal. The response of the system due to error signal is given by convolution of error signal and impulse response. The variance of response of the system for error signal is called state output noise power.

14. What is meant by coefficient inaccuracy?

In digital computation the filter coefficients are represented in binary. With b-bit binary we can generate only 2b different binary numbers and they are called quantization levels. Any filter coefficient has to be fitted into any one of the quantizat6ion levels. Hence the filter coefficients are quantized to represent in binary and the quantizatiion introduces errors in filter coefficients. Therefore the coefficients cannot be accurately represented in a digital system and this problem is referred to as coefficient inaccuracy.

15. How the digital filter is affected by quantization of filter coefficients?

The quantization of the filter coefficients will modify the value of poles & zeros and so the location of poles and zeros will be shifted from the desired location. This will create deviations in the frequency response of the system. Hence the resultant filter will have a frequency response different from that of the filter with unquantized coefficients.

16. How the sensitivity of frequency response to quantization of filter coefficients is minimized?

The sensitivity of the filter frequency response to quantization of the filter coefficients is minimized by realizing the filter having a large number of poles and zeros as an interconnection of second order sections. Hence the filter can be realized in cascade or parallel form, in which the basic buildings blocks are first order and second order sections.

17. What is meant by product quantization error?

In digital computations, the output of multipliers i.e., the product are quantized to finite word length in order to store them in registers and to be used in subsequent calculations. The error due to the quantization of the output of multiplier is referred to as product quantization error.

18. Why rounding is preferred for quantizing the product?

In digital system rounding due to the following desirable characteristic of rounding performs the product quantization

1. The rounding error is independent of the type of arithmetic

2. The mean value of rounding error signal is zero.

3. The variance of the rounding error signal is least.

19. Define noise transfer function (NTF)?

The Noise Transfer Function is defined as the transfer function from the noise source to the filter output. The NTF depends on the structure of the digital networks.

20. What are the assumptions made regarding the statistical independence of the various noise sources in the digital filter?

The assumptions made regarding the statistical independence of the noise sources are,

1. Any two different samples from the same noise source are uncorrelated.
2. Any two different noise source, when considered, as random processes are uncorrelated.
3. Each noise source is uncorrelated with the input sequence.

21. What are limit cycles?

In recursive systems when the input is zero or some nonzero constant value, the nonlinearities die to finite precision arithmetic operations may cause periodic oscillations in the output. These oscillations are called limit cycles.

22. What are the two types of limit cycles?

The two types of limit cycles are zero input limit cycles and overflow limit cycles.

23. What is zero input limit cycles?

In recursive system, the product quantization may create periodic oscillations in the output. These oscillations are called limit cycles. If the system output enters a limit cycles, it will continue to remain in limit cycles even when the input is made zero. Hence these limit cycles are also called zero input limit cycles.

24. What is dead band?

In a limit cycle the amplitudes of the output are confined to a range of values, which is called dead band of the filter.

25. How the system output cam be brought out of limit cycles?

The system output can be brought out of limit cycle by applying an input of large magnitude, which is sufficient to drive the system out of limit cycle.

26. What is saturation arithmetic?

In saturation arithmetic when the result of an arithmetic operation exceeds the dynamic range of number system, then the result is set to maximum or minimum possible value. If the upper limit is exceeded then the result is set to maximum possible value. If the lower limit is exceeded then the r4esult is set to minimum possible value.

27. What is overflow limit cycle?

In fixed point addition the overflow occurs when the sum exceeds the finite word length of the register used to store the sum. The overflow in addition may lead to oscillations in the output which is called overflow limit cycles.

28. How overflow limit cycles can be eliminated?

The overflow limit cycles can be eliminated either by using saturation arithmetic or by scaling the input signal to the adder.

29. What is the drawback in saturation arithmetic?

The saturation arithmetic introduces nonlinearity in the adder which creates signal distortion.

30. What are the factors that may be considered when selecting a DSP processor for an application?

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

31. What is meant by pipelining?

A pipeline is the continuous and somewhat overlapped movement of instruction to the processor or in the arithmetic steps taken by the processor to perform an instruction. With pipelining, the computer architecture allows the next instructions to be fetched while the processor is performing arithmetic operations, holding them in a buffer close to the processor until each instruction operation can be performed. The staging of instruction fetching is continuous. The result is an increase in the number of instructions that can be performed during a given time period.

32. What is the principle features of the Harvard Architecture?

The Harvard architecture has two separate memories for their instructions and data. It is capable of simultaneous reading an instruction code and reading or writing a memory or peripheral.

33. Differentiate between von Neumann and Harvard architecture?

Harvard Architecture

1.Von-Neumann Architecture Separate memories for program and data. It shares same memory for program and data.

2.The speed of execution in Harvard architecture is high. The speed of execution is increased by pipelining

3.In this architecture having a common interval address and data bus.It is having a separate interval address and data bus.

4.It is not suitable for DSP processors. It is normally used for Harvard architecture.

34. Give the digital signal processing application with the TMS 320 family.

- DSP processors should have circular buffers to support circular shift operations.
- The DSP processor should be able to perform multiply and accumulate operations very fast.
- DSP processors should have multiple pointers to support multiple operands jumps and shifts.

35. What is the advantage of Harvard architecture of TMS 320 series?

- It shares same memory for program and data
- The speed of execution is increased by pipelining
- It is having a separate interval address and data bus.
- It is normally used for Harvard architecture

36. What are the desirable features of DSP Processors?

DSP processors should have multiple registers so that data exchange from register to register is fast.DSP operations require multiple operands simultaneously. Hence DSP processor should have multiple operand fetch capacity.DSP processors should have circular buffers to support circular shift operations. The DSP processor should be able to perform multiply and accumulate operations very fast.DSP processors should have multiple pointers to support multiple operands jumps and shifts. Multi processing ability.

37. What are the different types of DSP Architecture?

Von-Neumann Architecture
Harvard Architecture
Modified Harvard Architecture
VLIW Architecture

38. Define MAC unit?

The dedicated hardware unit is called MAC. It is called multiplier-accumulator. It is one of the computational unit in processor. The complete MAC operation is executed in one clock cycle.

The DSP processors have a special instruction called MACD. This means multiply accumulate with data shift.

39. Mention the Addressing modes in DSP processors.

- Short immediate addressing
- Short Direct Addressing
- Memory-mapped Addressing
- Indirect Addressing
- Bit reversed addressing mode
- Circular addressing

40. State the features f TMS3205C5x series of DSP processors.

♣Powerful 16 bit CPU
♣TDM port
♣16X16 bit multiplies / Add operations can be performed in single cycle.
♣Full duplex synchronous serial port for coder / decoder interface.
♣On-chip scan based emulation logic.
♣Boundary scan
♣Low power dissipation
♣IEEE standard text access ports

41. Define Parallel logic unit?

It executes logic operations on the data without affecting the contents of ACC. PLU provides bit manipulation which can be used to set, clear, test or toggle bits in data memory control or status registers.

42. Define scaling shifter?

The scaling shifter has a 16 bit input connected to the data bus and 32 – bit output connected to the ALU. The scaling shifter produces a left shift of 0 to 16 bits on the input data. The other shifters perform numerical scaling, bit extraction, extended precision arithmetic and overflow prevention.

43. Define ARAU in TMS320C5X processor?

ARAU meant Auxiliary register and auxiliary register arithmetic unit. These register are used for temporary data storage. The auxiliary register file is connected to the auxiliary register arithmetic unit. The contents of the auxiliary register can be ARAU helps to speed up the operations of CALU.16. What are the Interrupts available in TMS320C5X processors?

It has four general purpose interrupts.
•INT4
•INT1
•RS (Reset)
•NMI (Non – Maskable interrupt)

44. What are the addressing modes available in TMS320C5X processors?
•Direct
•Indirect
•Immediate
•Register
•Memory mapped
•Circular Addressing

45. What are the different buses of TMS320C5X and their functions?
The C5X architecture has four buses and their functions are as follows:
Program bus (PB):It carries the instruction code and immediate operands from program memory space to the CPU. Program address bus (PAB):It provides addresses to program memory space for both reads and writes. Data read bus (DB): It interconnects various elements of the CPU to data memory space. Data read address bus (DAB):It provides the address to access the data memory space.

46. List the on-chip peripherals in 5X.

The C5X DSP on-chip peripherals available are as follows:
1. Clock Generator
2. Hardware Timer
3. Software-Programmable Wait-State Generators
4. Parallel I/O Ports
5. Host Port Interface (HPI)
6. Serial Port
7. Buffered Serial Port (BSP)
8. Time-Division Multiplexed (TDM) Serial Port
9. User-Maskable Interrupts

47. 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.

48. What are the different stages in pipeling?

(i)The fetch phase, (ii)The decode phase, (iii) Memory read phase, (iv) The execute phase.

49. Why do we need DSP processors?

Use a DSP processor when the following are required:
Cost saving
Smaller size
Low power consumption
Processing of many “high” frequency signals in real-time

50. What are the basic instruction features of DSP?
Arithmetic operations such as add, subtract and multiply
Logic operations such as AND,OR,XOR and NOT
Multiply and accumulate (MAC) operation
Signal scaling operations for scaling the signal before and/or after digital signal processing. It is important that dedicated high-speed hardware be provided to carry out these operations.

51. What are the basic hardware features of DSP?
- On-chip registers for storage of intermediate results
- On-chip memories for signal samples (RAM)
- On-chip program memory for programs and fixed data such as filter coefficients (ROM)

52. What are the advantages and disadvantages of VLIW architecture?
Advantages of VLIW architectures

Increased performance
Better compiler targets
Potentially scalable
Disadvantages of VLIW architectures
Increased memory use High power consumption.

UNIT – V DSP APPLICATIONS

1. What is decimation?

"Decimation" is the process of reducing the sampling rate. In practice, this usually implies lowpass-filtering a signal, then throwing away some of its samples.

2. What is the "decimation factor"?

The decimation factor is simply the ratio of the input rate to the output rate. It is usually symbolized by "M", so input rate / output rate=M.

3. Why decimate?

The most immediate reason to decimate is simply to reduce the sampling rate at the output of one system so a system operating at a lower sampling rate can input the signal. But a much more common motivation for decimation is to reduce the *cost* of processing: the *calculation* and/or *memory* required to implement a DSP system generally is proportional to the sampling rate, so the use of a lower sampling rate usually results in a cheaper implementation.

4. Which signals can be downsampled?

A signal can be downsampled (without doing any filtering) whenever it is "oversampled", that is, when a sampling rate was used that was greater than the Nyquist criteria required. Specifically, the signal's highest frequency must be less than half the post-decimation sampling rate. (This just boils down to applying the Nyquist criteria to the input signal, relative to the new sampling rate.)

5. What is sub band coding?

In signal processing, sub-band coding (SBC) is any form of transform coding that breaks a signal into a number of different frequency bands, typically by using a fast Fourier transform, and encodes each one independently. This decomposition is often the first step in data compression for audio and video signals.

6. State the various applications of DSP.

Applications of DSP include audio signal processing, audio compression, digital image processing, video compression, speech processing, speech recognition, digital communications, digital synthesizers, radar, sonar, financial signal processing, seismology and biomedicine. Specific examples include speech coding and transmission in digital mobile phones, room correction of sound in hi-fi and sound reinforcement applications, weather forecasting, economic forecasting, seismic data processing, analysis and control of industrial processes, medical maging such as CAT scans and MRI, MP3 compression, computer graphics, image manipulation, audio crossovers and equalization, and audio effects units.

7. What is multirate digital signal processing?

The processing of a discrete time signal at different sampling rates in different parts of a system is called multirate DSP.

8. What is echo cancellation?

Echo cancellation is a method used in telephony and telecommunication to improve voice quality by preventing echos from being captured or created, or possibly removing it in post-processing. The aim is filter the echo that is either produced by acoustic means or a hybrid echo produced by line echo, electrical reflections or impedance mismatch

9. Mention two applications of multirate signal processing.

•Phase shifter
•Transmultiplexers
•Vocoder

10. What is an anti-imaging filter?

The filter which is used to remove the image spectra is known as anti-imaging filter.

11. What is trans multiplexers?

Trans multiplexers is used to convert frequency division multiplexed signals into time division multiplexed signals and vice versa.

12. Mention the two stages of musical sound processing.

First stage: the sound from the singer or the sound from the instrument is recorded on a single track of multi track tape. Second stage: the special audio effects are added to this sound. The special audio effects can be generated by DSP.

13.What are the steps to be followed for the reproduction of the recorded signal?

•Decoding and demodulation

•Error correction and demultiplexing.

14. Define interpolation.

The process of increasing the sampling rate by a factor I is known as interpolation.

15.Define aliasing.

Aliasing refers to an effect that causes different signals to become indistinguishable when sampled. It also refers to the distortion when the signal reconstructed from samples is different from the original continuous signal.

16. What do you mean by adaptive signal processing?

An adaptive filter has an adaptation algorithm, that is meant to monitor the environment and vary the filter transfer function accordingly. The algorithm starts from a set of initial conditions, that may correspond to complete ignorance about the environment, and, based in the actual signals received, attempts to find the optimum filter design. In a stationary environment, the filter is expected to converge, to the Wiener filter. In a nonstationary environment, the filter is expected to track time variations and vary its filter coefficients accordingly.

17. What are the basic operations involved in adaptive signal processing?

The basic operation now involves two processes:

1. a filtering process, which produces an output signal in response to a given input signal.

2. an adaptation process, which aims to adjust the filter parameters (filter transfer function) to the (possibly time-varying) environment. The adaptation is steered by an error signal that indicates how well the filter output matches some desired response. Examples are given in the next section. Often, the square value of the error signal is used as the optimization criterion.

18. What are the two methods used for sampling rate conversion?

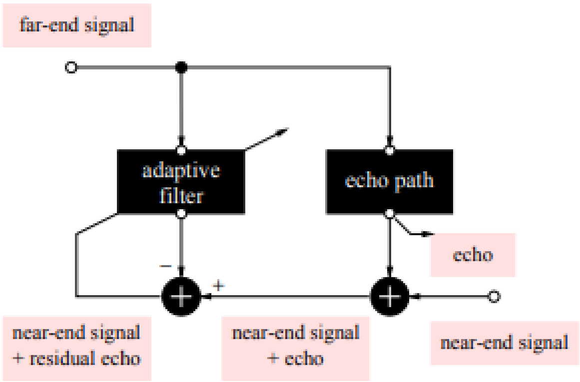
First method:

The digital signal is converted into analog signal by using DAC. Then analog signal is converted into digital signal using ADC.

Second method:

Sampling rate conversion is performed in digital domain.

19. Draw the block diagram for adaptive echo cancellation.



20. List the applications of acoustic noise reduction.

Acoustic noise reduction is particularly useful when low bit rate coding schemes (e.g. LPC) are used for the digital representation of the speech signal. Such coders are very sensitive to the presence of background noise, often leading to unintelligible digitized speech. It is also for useful for main electricity interference cancellation (i.e. removal of 50-60 Hz sinewaves), where the reference signal is taken from a wall outlet.