QUESTION BANK

Name of the Department : **Electronics and Communication Engineering**

Subject Code & Name : **EC8553 & Discrete-Time Signal Processing**

Year & Semester : **III & V**

UNIT I DISCRETE FOURIER TRANSFORM

PART-A

1. **How many multiplications and additions are required to compute N point DFT using radix-2 FFT?**
   The number of multiplication and additions required to compute N-point DFT using radix-2 FFT are
   Additions: $N\log_2 N$; Multiplication: $N/2\log_2 N$

2. **Why the computations in FFT algorithm is said to be in place?**
   OR
   What do you mean by in-place computation in FFT.
   The (A,B) are calculated from (a,b). Hence (A,B) can be stored in place of (a, b) since (a,b) are not required further. This is called in place computation. It reduces the number of memory locations.

3. **What is the relationship between Fourier transform and DFT?**

S.No	Fourier transform	DFT
1	Converts the signal from time domain to frequency domain.	DFT can be evaluated using fast algorithms.
2	Fourier transform is mainly used for nonperiodic signals.	Discrete Fourier transform is mainly used for nonperiodic signals.
3	Continous function of w	Discrete frequency frequency spectrum
4	Sampling is performed only in time domain	Obtained by performing sampling operation in both the time and frequency domains

4. What is twiddle factor?

The complex valued phase factor W_N is called twiddle factor $W_N = e^{-j2\pi/N}$

5. State and prove periodicity property of DFT.

If X(k) is N-point DFT of a finite duration sequence x(n)

Then

x(n+N)=x(n) for all n

X(k+N)=X(k) for all k

6. What is relation between DTFT and DFT? [

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

7. Compare Radix-2 DIT, DIF FFT algorithm.

S.No	DIT	DIF
1	The input is bit reversed	The input is in natural order.
2	The output is in natural order	The output is bit reversed.
3	In the Butterfly diagram after the multiplication only we have to perform add-subtract operation	In the Butterfly diagram, the complex multiplication take place after the ass-subtract operation.

8. Difference between Analog and Digital signal processing.

S.No	Analog Signal Processing	Digital Signal Processing
1	It has less flexibility	It has more flexibility
2	Accuracy is not good	Accuracy is high
3	It has high cost for processing	It has lower cost for processing
4	ADC and DAC converters are not required	ADC and DAC converters are required

9. Classify the different Discrete Time Signal.

- Energy and power signals

- Periodic and aperiodic signals
- Even and odd signals
- Causal and non-causal signals

10. Define Energy and power Signal .

The energy E of asignal x(t) and x(n) is defined as ,

$E=\int |x(t)|^2$ dt for continous time signal

$E=\Sigma |x(n)|^2$ for discrete time signal

The energy of a signal can be finite or infinite,. If E is finite then the signal is an energy signal.

The power P of a signal x(t) and x(n) is defined as

$P= \lim 1/T \int |x(t)|^2$ dt for continous time signal

$P= \lim 1/2N+1 \ \Sigma |x(n)|^2$ for discrete time signal

The power of a signal can be finite or zero. If P is finite, then the signal is power signal.

11. **Define periodic and aperiodic signal**.

A signal x(n) is periodic with period N (N> 0) if and only if x(n+N)=x(n) for all n.

If there is no value of N then signal is called non-periodic

12. **Classify discrete time systems**.[D]

- Static and dynamic systems
- Causal and non-causal systems
- Linear and non-linear systems
- Time-variant and Time-invariant systems
- Stable and unstable systems
- FIR and IIR systems

13. What are the advantages of FFT over DFTs?

- FFT are the algorithms used to compute DFT fast.
- FFT algorithms are computationally efficient than direct computation of DFT.
- FFT algorithms exploit periodicity and symmetry properties of DFT.

14. What do you understand by the terms: Signal and Signal processing.

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

Signal processing is an operation that changes the characteristics of a signal. These characteristics include the amplitude, shape, phase and frequency content of a signal.

15. Define symmetric and antisymmetric signals.

A real valued signal x(n) is called symmetric if x(-n)=x(n).

On the other hand, a signal x(n) is called antisymmetric if x(-n)=-x(n).

16. What are the different types of signal representation?

- Graphical representation
- Functional representation
- Tabular representation
- Sequence representation

17. What is the property of shift-invariant system? (OR) What is a time-invariant system? (OR) What is a shift-invariant system?

If the input-output relation of a system does not vary with time, the system is said to be time-invariant or sift-invariant.

If the output signal of a system shifts k units of time upon delaying the input signal by k units the system under consideration is a time-invariant system.

Ex: $y(n)=x(n)+x(n-1)$

18. Define DTFT pair.

$x(n) = 1/2\pi \int X(e^{jw})\, e^{jwn}\, dw$

$X(e^{jw}) = \Sigma x(n)\, e^{-jwn}$

19. What is aliasing effect?

Let us consider a band limited signal x(t) having no frequency component for $|\Omega| > \Omega_m$. If we sample the signal x(t) with a sampling frequency $F<2f_m$, the periodic continuation of $X(j\Omega)$ results in spectral overlap. In the case the spectrum $X(j\Omega)$ cannot be recovered using a low pass filter. This effect is known as aliasing effect.

20. State sampling theorem.

A band limited continuous time signal with higher frequency f_m Hertz, can be uniquely recovered from its samples provided the sampling rate $F \geq 2f_m$ samples per second.

21. 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), we have to add (N-L) Zeros to the sequence x(n). This is known as Zero padding. The uses of padding a sequence with Zero are

- We can get better display of the frequency spectrum
- With zero padding the DFT can be used in linear filtering.

22. State the difference between overlap save method and overlap add method.

	Overlap save method	Overlap add method
1	In this method the size of the input data block is N=L+M-1	In this method the size of the input data block is L.
2	Each data block consists of the last M-1 data points of the previous data block followed by	Each data block is L points and we append M-1 zeros to compute N-point DFT

	L new data points.	
3	To form the output sequence the first M-1 data points are discarded in each output block and the remaining data are fitted together.	To form the output sequence the last M-1 points from each output block is added to the first (m-1) points of the succeeding block.

23. Why FFT is needed?

The direct evaluation of DFT using the formula $X(k)=\Sigma\ x(n)\ e^{-j2\pi nk/N}$ requires N2 complex multiplications and N(N-1) complex additions. Thus for reasonably large values of N direct evaluation of the DFT requires an inordinate amount of computation. By using FFT algorithms the number of computations can be reduced.

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

The number of complex multiplication required using direct computation is

$N^2=64^2=4096$

The number of complex multiplications required using fft is

$N/2\ \log_2 N=64/2\ \log_2 64=192.$

Speed improvement factor =4096/192=21.33

25. What is FFT?

The fast fourier transform is an algorithm used to compute the DFT. It makes use of the symmetry and periodicity properties of twiddle factor w^k_N to effectively reduce the DFT computation time. It is based on the fundamental principle of decomposing the computation of 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 DET of approximately 64 and 205 for 256 point and 1024 point transforms respectively.

26. What is meant by radix-2 FFT?

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

27. List any four properties of DFT.

- Periodicity
- Linearity
- Time reversal of a sequence
- Circular time shifting of a sequence.

28. What are the applications of FFT algorithm?

- Linear filtering
- Correlation
- Spectral analysis

29. What is meant by bit-reversal operation?

In DIT algorithm we can find that for the output sequence to be in a natural order and the input sequence has to be stored in a shuffled order. When N is a power of 2, the input sequence must be stored in bit reversal order for the output to be computed in a natural order.

30. What is meant by Decimation in time algorithm?

DIT algorithm is used to calculate the DFT of N-point sequence. It break the N-point sequence into two sequence as $x_e(n)$ and $x_0(n)$ which have the even and odd values of x(n). The N/2 point DFTs of these two sequence are evaluated and combination of N/4 point DFTs. This process will be performed continuously until we are left with 2-point DFT. This algorithm is called decimation in time because the sequence x(n) is often split into smaller subsequences.

31. What is meant by Decimation in frequency algorithm?

DIF splits the DFT X(k) into odd and even number of samples that is why the name decimation in frequency.

PART-B

1. Consider the length-8sequnce defined for $0 \leq n \leq 8$
   $x(n) = \{1,2,-3,0,1,-1,4,2\}$
   with a 8-point DFT. Evaluate the following functions of X(k) without computing DFT. (i) X(0) (ii) X(4) (iii) $\sum_{k=0}^{7} X(k)$ (iv) $\sum e^{-j3\pi/4} X(k)$ (v) $\sum |X(k)|^2$.

2. State and prove any four properties of DFT.

3. Perform circular convolution of the following sequences $x_1(n) = \{1, 1, 2, 1\}$; $x_2(n) = \{1,2,3,4\}$.

4. Mention the differences and similarities between DIT and DIF algorithms

5. Compute 4 point DFT of a sequence $x(n) = \{0, 1, 2, 3\}$ using DIF and DIT algorithms.
6. Find the 8 point DFT of the sequence $x(n) = \{2, 2, 2, 2, 1, 1, 1, 1,\}$ using Decimation in Time FFT algorithm.
7. Determine the circular convolution of the sequence $x_1(n) = \{1, 2, 3, 1, 1, 2, 3, 1\}$ and $x_2(n) = \{4, 3, 2, 2, 2, 2, 3, 4\}$ using DFT and IDFT.

8. Compute the DFT for the sequence {1,2,3,4,4,3,2,1}. Using radix-2 DIF-FFT algorithm.
9. In an LTI system the input x(n)={1, 1, 2, 1} and the impulse response h(n) ={ 1, 2, 3, 4}. Perform the circular convolution using DFT and IDFT.

10. Derive radix 2-DIT FFT algorithm and obtain DFT of the sequence x(n) ={1, 2, 3, 4, 4, 3, 2, 1} using DIT algorithm.

11. Compute IDFT of the sequence X(k) ={ 7, -0.707 -j0.707, -j, 0.707-j0.707, 1, 0.707+j0.707, j, -0.707+j0.707} using DIF algorithm.

12. Perform the linear convolution of finite duration sequences h(n)= {1, 2} and X(n) = {1 , 2, -1, 2, 3, -2, -3, -1, 1, 2, -1} by overlap save method.

13. Compute the 8 point circular convolution
    $X_1(n)$= { 1, 1, 1, 1, 0, 0, 0, 0}
    $X_2(n) = \sin 3\pi n/8$, $0 \le n \le 7$ using matrix method.

14. Perform linear convolution of finite duration sequences h(n)={1, 1, 2, 1} and x(n)= {1, -1, 1, 2, 1, 0, 1, -4, 3, 2, 1, 0, 1, 1} by overlap-add method and overlap save method.

15. Given the sequence $x_1(n)$= {1, 2, 3, 4}; $x_2(n)$= {1, 1, 2, 2}.Find $x_3(n)$ such that $x_3(k)= x_1(k)\, x_2(k)$.

16. Find the circular convolution of the two sequence $x_1(n)$= {1, 2, 2, 1}; $x_2(n)$= {1, 2, 3, 1}. Using (a) concentric circle method (b) matrix method

17. Find the 8-point DFT of the sequence x(n) = {1, 1, 1, 1, 1, 0, 0, 0} usinf DIT-FFT algorithm. (13)

18. Compute the eight point DFT of the sequence
    $$X(n)= \begin{cases} 1 & 0 \le n \le 7 \\ 0 & \text{otherwise} \end{cases}$$
    by using DIF algorithm.

19. Compute the eight point DFT of the sequence x(n) ={ 0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0} using the inplace radix-2 DIT algorithm.

20. Evaluate and compare the 8-point for the following sequence using DIT-FFT algorithm.

$X_1(n)= \{1 \quad \text{for } -3 \le n \le 3$
$\quad\quad 0 \quad \text{otherwise}$

21. Find the 8-point DFT of the given sequence x(n) ={0, 1, 2, 3, 4, 5, 6, 7} Using DIF,radix-2, FFT algorithm.

UNIT-II INFINITE IMPULSE RESPONSE FILTERS

PART-A

1. **What are the properties of bilinear transformation?**
   - The mapping for the bilinear transformation is a one to one mapping
   - The jΩ-axis maps on to the unit circle |z|=1 the left half of the s-plane maps to the interior of the unit circle |z|=1 and the right half of the S-plane maps on to the exterior of the unit circle |z|=1.

2. **What is meant by frequency warping? (or) What is known as warping effect?**
   The relationship or relation between the analog and digital frequencies in bilinear transformation is given by
   $\Omega=2/T\tan\omega/2$
   For smaller values of ω, there exist linear relationship between ω and Ω. But for large values of ω, the relationship is non-linear. This non linear introduces distortion in the frequency axis. This known as warping effort.

3. **What are the methods used for digitizing the analog filter into a digital filter?**
   - Map the desired digital filter specifications into those for an equivalent analog filter.
   - Derive the analog transfer function for the analog prototype.
   - Transform the transfer function of the analog prototype into an equivalent digital filter transfer function.
   -

4. **List the different types of filters based on frequency response? (OR) What are the types of filters based on frequency response?**
   - Low pass filters
   - High pass filters
   - Band pass filters
   - Band stop filters

5. **Discuss the need for prewarping. (or) What is known as prewarping?**
   The effort of the non linear compression at high frequencies can be compensated. When the desired magnitude response is piece-wise constant over frequency this

compression can be compensated by introducing a suitable prescaling or prewarping the critical frequencies by using formula.

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

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

- The analog filter transfer function $H_a(s)$ should be a rational function of a and coefficient of s should be real.
- The poles should lie on left half of s-plane.
- The number of Zeros should be less than or equal to number of poles.

7. Why impulse invariant method is not preferred in the design of IIR filter other than lowpass filter?

In impulse invariance method the mapping from s-plane to z-plane is many to one . all the poles in the s-plane between the intervals $(2k-1)\pi/T$ to $(2k+1)\pi/T$ map into the entire z-plane. Thus infinite number of poles that map to the same location in the z-plane producing aliasing effect. Due to spectrum aliasing the impulse invariance method is inappropriate for designing highpass filter. That is why the impulse invariance method is not preferred in the design of IIR filter other than lowpass filter.

8. Give any two properties of Butterworth lowpass filter.

- The magnitude response of the Butterworth filter decreases monotonically as the frequency Ω increases from 0 to ∞
- The magnitude response of the Butterworth filter closely approximates the ideal response as the order N increases.
- The poles of the Butterworth filter lie on a circle.

9. What are the properties of chebyshev filter?

- The magnitude response of the chebyshev filter exhibits ripple either in passband or in stopband according to type.
- The poles of the chebyshev filter lie on an ellipse.

10. What is bilinear transformation?

The bilinear transformation is a mapping that transforms the left half of s-plane into the unit circle in the z-plane only once thus avoiding aliasing of frequency components.

The mapping from the s-plane to the z-plane in bilinear transformation is

$S=2/T\ [1-z^{-1}/1+z^{-1}]$

11. What are the advantage and disadvantage of bilinear transformation?

Advantages:

- The bilinear transformation provides one to one mapping
- Stable continuous systems can be mapped into realizable stable digital systems.
- There is no aliasing.

Disadvantages:

- The mapping is highly non-linear producing frequency compression at high frequencies.
- Neither the impulse response nor the phase response of the analog filter is preserved in a digital filter obtained by bilinear transformation.

12. List the various forms of realizations of IIR system.

- Direct form-I
- Direct form-II
- Cascade realization
- Parallel form realization
- Lattice realization

13. Mention two transformations to digitize an analog filter.

- Bilinear transformation
- Impulse invariant transformation

14. Compare FIR and IIR filters.

S.No	FIR	IIR
1	Finite impulse response	Infinite impulse response
2	It has finite duration unit sample response.	It has infinite duration unit sample response.
3	It depends on present and past input only	It depends on present and past inputs as well as its outputs.

15. Distinguish between the frequency response of cheybyshev type I and type II filter.

	Chebyshev filter Type-I	Chebyshev filter type-II
1	Filters are all pole filters that exhibits equiripple behavior in the pass band and a monotonic characteristics in the stopband.	Filter contains both poles and zeros and exhibits a monotonic behavior in the pass band and an equiripple behavior in the stopband.

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

The poles of the Butterworth filter is given by

$S_k=e^{j\phi k}$ where k=1,2,……N

$\phi k = \pi/2 + (2k-1)\pi/2N$

17. Compare analog and digital filters.

	Analog filter	Digital filter
1	Frequency response can be changed by changing the components	Frequency response can be changed by changing the filter coefficients.
2	It process and generates analog output	It process and generates digital output.

18. What are properties of impulse invariance transformation?

- All poles in the right half of the s-plane map to digital poles outside the unit circle.
- Although the jΩ axis is mapped into unit circle it is not one to one mapping rather it is many to one mapping.

19. What is the advantage of cascade realization?

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

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

Based on impulse response the filters are of two types

1. IIR filter
2. FIR filter

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

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

21. State the advantage of direct form I structure over direct form II structure.

In direct form _structure, the number of memory locations required is less than that of direct form structure.

22. What do you understand by backward difference?

One of the simplest method for converting an analog filter into a digital filter is to approximate the differential equation by an equivalent difference equation.

d/dt y(t)=y(nT)-y(nT-T)

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

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

H(Z) =H(S)|S=(1-Z-1)/T

The above mapping has the following characteristics

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

• The right half of S-plane maps into the region outside the circle of radius ½ in the Zplane.
• The j Ω-axis maps onto the perimeter of the circle of radius ½ in the Zplane.

PART-B

1. Given the specifications $\alpha_p = 3dB$; $\alpha_s = 16dB$; f_p=1KHz and f_s =2KHz. Determine the order of the filter using chebyshev approximation. Find H(s).

2. Using the bilinear transform design a high pass filter, monotonic in pass band with cutoff frequency of 1000Hz and down 10dB at 350Hz. The sampling frequency is 5000Hz.

3. Design an analog Butterworth filter for a given specifications.
$0.9 \le |H(j\Omega)| \le 1$ for $0 \le \Omega \le 0.2\pi$.
$|H(j\Omega)| \le 0.2$ for $0.4\pi \le \Omega \le \pi$.

4. Apply impulse invariant method and find H(z) for H(s) = s+a / $(s+a)^2 + b^2$

5. Apply bilinear transformation to H(s) =2/(s+1)(s+2) with T=1 sec and find H(z).

6. Enumerate the steps for IIR filter design by impulse invariance with an example.

7. Analyze the design of discrete time IIR filter from analog filter.

8. Design a digital Butterworth filter with the following specifications.
$0.707 \le |H(e^{j\omega})| \le 1$ for $0 \le \omega \le 0.5\pi$.
$|H(e^{j\omega})| \le 0.2$ for $0.75\pi \le \omega \le \pi$. Determine system function H(z) for a Butterworth filter using Bilinear transformation.

9. Determine the system function of the lowest order digital chebyshev filter with the following specifications, 3db ripple in the pass band $0 \le \omega \le 0.2\pi$ and 25db attenuation in the stop band $0.45\pi \le \omega \le \pi$.

10. Design a third order Butterworth digital filter using impulse invariant technique. Assume sampling period T=1 sec.

11. If $H_a(S)$ = 1/(S+1)(S+2), find the corresponding H(z) using impulse invariant method for sampling frequency of 5 samples/Second.

12. Obtain the direct form-I realization for the system described by difference equation (i) $y(n) = 0.5y(n-1) - 0.25y(n-2) + x(n) + 0.4x(n-1)$
    (ii) $y(n) = 2y(n-1) + 3y(n-2) + x(n) + 2x(n-1) + 3x(n-2)$

13. Obtain direct form-I and direct form II realization of the LTI system governed by the equation
    $y(n) = -3/8y(n-1) + 3/32y(n-2) + 1/64\, y(n-3) + x(n) + 3x(n-1)$

14. Convert the following analog filter with transfer function
    $H_a(s) = s+0.2/(s+0.2)^2 + 16$ using bilinear transformation method

15. Design a digital chebyshev filter to meet the constraints
    $1/\sqrt{2} \leq |H(e^{j\omega})| \leq 1$ for $0 \leq \omega \leq 0.2\pi$.
    $0 \leq |H(e^{j\omega})| \leq 0.1$ for $0.5\pi \leq \omega \leq \pi$.
    By using bilinear transformation and assume sampling period T=1sec

16. Realize the following IIR system by cascade and parallel forms.
    $y(n) + 1/4y(n-1) – 1/8y(n-2) = x(n) - 2x(n-1) + x(n-2)$

17. 17. Design a Butterworth filter using the impulse invariance method for the following specifications.
    $0.8 \leq |H(e^{j\omega})| \leq 1$ for $0 \leq \omega \leq 0.2\pi$
    $|H(e^{j\omega})| \leq 0.2$ for $0.6\pi \leq \omega \leq \pi$.

UNIT-III FINITE IMPULSE RESPONSE FILTERS

PART-A

1. **What is the necessary and sufficient condition for linear phase characteristic in FIR filter?**
   The necessary and sufficient condition for linear phase characteristic in FIR filter is the impulse response h(n) of the system should have the symmetrical property.
   $h(n) = h(N-1-n)$
2. **Write the steps involved in FIR filter design.**
   - From the given frequency response calculate required order of the filter.
   - From the order and desired frequency response calculate desired unit sample response $h_d(n)$.
   - From the attenuation characteristics select suitable window function $\omega(n)$.
   - Calculate $h(n) = h_d(n).\ \omega(n)$.

3. What is Gibbs phenomenon?

One possible way of finding an FIR filter that approximates $H(e^{j\omega})$ would be to truncate the infinite fourier series at n=±[N-1/2]. Abrupt truncation of the series will lead to oscillation both in pass band and in stop band. This phenomenon is known as Gibbs phenomenon.

4. Compare Hamming window with Kaiser window.

S.No	Hamming window	Kaiser window
1	The main lobe width is equal to 8π/N and the peak side lobe level is -41 Db	The main lobe width the peak side lobe level can be varied by varying the parameter α and N.
2	The low pass FIR filter designed will have first side lobe peak of -53dB	The side lobe peak can be varied by varying the parameter α

5. What do you understand by linear phase response? (OR) What do you meant by linear phase response?

The phase response of the type

$<H(\omega) = K\omega$, K is constant

ω is called linear phase response. The linear phase filter does not alter the shape of the original signal. In many case a linear phase characteristic is required throughout the pass band of the filter to preserve the shape of a given signal within the pass band.

6. What are the desirable characteristics of windows?

- Central lobe of the frequency response of the window should contain most of the energy and it should be narrow.
- The highest side lobe level of the frequency response should be small.
- The side lobe of the frequency response should decrease in energy rapidly as ω tends to π.

7. What are the advantage and disadvantage of FIR filters? (OR) What are the desirable and undesirable features of FIR filters? (OR) List out the advantage of FIR filters.

Advantages:

- FIR filters have exact linear phase.
- They are always stable.
- FIR filters can be realized in both recursive and non-recursive structure.
- They design methods are generally linear.

Disadvantage:

- Large storage requirements needed.

- Powerful computational facilities required for the implementation.
- For the same filter specification the order of FIR filter design can be as high as 5 to 10 times that of an IIR design.

8. What is reason that FIR filter is always stable?
FIR filter is always stable because all its poles are at the origin.

9. State the conduction 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 is stable if its impulse response is absolutely summable.
$\Sigma |h(n)| < \infty$

10. What are the different types of digital filters based on impulse response?
Based on impulse response, digital filters are classified into two types.
- Finite duration unit pulse response (FIR) filters.
- Infinite duration unit pulse response (IIR) filters.

In the FIR system the impulse response sequence is of finite duration
The IIR system has an infinite number of non-zero terms.

11. What are the different types of filters based on frequency response?
- Low pass filter
- High pass filter
- Band pass filter
- Band reject filter

12. Distinguish between FIR and IIR filters.

	FIR filter	IIR filter
1	These filters can be easily designed to have perfectly linear phase.	These filters do not have linear phase.
2	FIR filters can be realized recursively and non-recursively	IIR filters are easily realized recursively.
3	Greater flexibility to control the shape of their magnitude response	Less flexibility, usually limited to specific kind of filters.

13. What are the techniques of designing FIR filters?
There are three well known methods for designing FIR filters with linear phase. These are
- Windows method
- Frequency sampling method
- Optimal or minimax design.

14. What are the properties of FIR filter?

- FIR filter is always stable
- A realizable filter can always be obtained
- FIR filter has a linear phase response.

15. For what type of filters frequency sampling method is suitable?

Frequency sampling method is suitable for filter that required the filtering only at particular frequencies. Such filters are narrowband frequency selective filters where only few samples of frequency response are non-zero.

16. Name different types of windowing functions.

- Hamming window
- Hanning window
- Rectangular window
- Kaiser window

17. What are the disadvantage of fourier series method?

In designing FIR filter using Fourier series method the infinite duration impulse response is truncated at n= ± [N-1/2]

18. What is the principle of designing FIR filter using windows? (OR) What is window and why it is necessary?

One possible way of obtaining FIR filter is to truncate the infinite Fourier series at n = ±[N-1/2] where N is the length of the desired sequence. But abrupt truncation of the fourier series results in oscillation in the passband and stopband. These oscillations are due to slow convergence of the fourier series.

19. How the zeros in FIR filter is located?

- Z_1 is a real zero with $|Z_1| < 1$. Then Z_1^{-1} is also a real zero and there are two zeros in this group.
- $Z_2 = -1$. Then $z_2^{-1}=z_2$ and this group contains only one zero.
- Z_3 is a complex zero with $|z_3| \neq 1$ then $Z_3^{-1} = z_3^*$ and there are two zeros in this group.
- Z_4 is a complex zero with $|z_4| \neq 1$. This group contains four zeros z_4, z_4^{-1}, z_4^*,$(z_4^*)^{-1}$

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

The conduction are

- Symmetric conduction h(n)=h(N-1-n)
- Antisymmetric conduction h(n) = -h(N-1-n)

21. List the different types of structures for realizing FIR system?

- Transversal structure
- Linear phase structure
- Polyphase realization of FIR filter.

22. What is the principle of designing FIR filter using frequency sampling method? (OR) What are the attractive aspects of frequency sampling design?

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

23. Define phase delay.

The phase delay is the ratio of phase angle with respect to frequency

$$\tau(\omega) = \theta(\omega)/\omega$$

For FIR filters the linear phase indicates constant phase delay.

24. Show that the filter with h[n]={-1 , 0,1} is a linear phase filter.

Here $h(0) = -1$, $h(1) = 0$ and $h(2) = 1$

For the linear phase filter,

$h(n) = \pm h(N-1-n)$

Since N=3, $h(n) = \pm h(3-1-n)$

$h(n) = \pm h(2-n)$

$h(0) = \pm h(2)$

$h(0) = -h(2)$. Hence this filter has linear phase.

PART-B

1. Using a rectangular window technique design a lowpass filter with pass band gain of unity, cutoff frequency of 1000 Hz and working at a sampling frequency of 5 KHz. The length of the impulse response should be 7.

2. Write the expression for the frequency response of Rectangular window and Hamming window and explain.

3. Determine the filter coefficients h(n) obtained by sampling

$$H_d(e^{j\omega}) = \begin{cases} e^{-j(N-1)\omega/2} & 0 \le |\omega| \le \pi/2 \\ 0 & \pi/2 \le |\omega| \le \pi \end{cases}$$

For N =7

4. Design a FIR filter with the following desired specifications, using Hanning window with N=5.
   $H_d(e^{j\omega}) = 0 , \quad -\pi/4 \le |\omega| \le \pi/4$
   $= e^{-j2\omega} , \quad -\pi/4 \le |\omega| \le \pi$

5. Explain the design procedure of FIR filter using frequency sampling method.

6. Design a HPF with the following frequency response
   $H_d(e^{j\omega}) = 1 , \quad \text{for} \quad \pi/4 \le |\omega| \le \pi$
   $= 0 , \quad \text{for} \quad |\omega| \le \pi/4$
   of length N=11 using Hanning window.

7. Determine the coefficients of a linear phase FIR filter of length N=15 which has a symmetric unit sample response and a frequency response that satisfies the conditions.
   $H(2\pi k/15) = 1$; for k = 0, 1, 2, 3.
   $= 0$; for k = 4, 5, 6, 7

8. Design an ideal BPF with a frequency response
   $H_\alpha(e^{j\omega}) = 1$, for $\pi/4 \le |\omega| \le 3\pi/4$
   0, otherwise
   Find the value of h(n) for N=11 and plot the frequency response.

9. Design a filter with $H_d(e^{j\omega}) = e^{-j3\omega} \quad -\pi/4 \le w \le \pi/4$
   $= 0, \quad \pi/4 \le |w| \le \pi$
   Using a Hamming window with N=7.

10. Design a high pass filter using hamming window with a cutoff frequency of 1.2radians/sec and N=9.

11. Design a low pass filter using rectangular window by taking samples of w(n) and with cut-off frequency of 1.2 rad/sec.

12. Realize $H(z) = 1 + 1/2\, z^{-1} + 1/8 z^{-2} + 3/4 z^{-3} + 1/8 z^{-4} + ½\, z^{-5} + z^{-6}$ with minimum number of multipliers.

13. Design an FIR lowpass filter satisfying the following specifications $\alpha_p \le 0.1$dB; $\alpha_s \ge 44.0$dB; $\omega_p = 20$rad/sec; $\omega_s = 30$rad/sec ; $\omega_{sp} = 100$rad/sec

14. Design an ideal lowpass filter with a frequency response
    $H_d(e^{j\omega}) = 1 , \quad \text{for} \quad -\pi/2 \le |\omega| \le \pi/2$
    $= 0 , \quad \text{for} \quad \pi/2 \le |\omega| \le \pi$
    Find the values of h(n) for N=11. Find H(z).

UNIT-4

FINITE WORD LEGTH EFFECTS

PART-A

1. **What is product quantization error? (OR) What is product roundoff error in digital signal processing?**
   Product quantization errors atrise at the output of a multiplier. Multiplication of a b-bit data with a b- bit coefficient results a product having 2b bits. Since a b – bit register is used the multiplier output must be rounded or truncated to b bits. Which produces an error. This error is known as product quantization error.

2. **What is meant by floating point representation?**
   In floating point arithmetic the set of signals to be handled is divided into blocks. Each block has the same value for the exponent. The arithmetic operations within the block use fixed point arithmetic and only one exponent per block is stored thus saving memory.

3. **What is meant by floating point arithmetic?**
   Floating point representation consists of mantissa M and exponent E. Floating point number us written as
   Binary floating point number: $M \times 2^E$
   Decimal floating point number : $M \times 10^E$

4. **Distinguish between fixed point arithmetic and floating point arithmetic.**

	Fixed point arithmetic	Floating point arithmetic
1	Fast operation	Slow operation
2	Relatively economical	More expensive because of costlier hardware
3	Small dynamic range	Increased dynamic range.
4	Roundoff errors occur only for addition	Roundoff error can occur with both addition and multiplication
5	Used in small computers	Used in larger, general purpose computers

5. **Why rounding is preferred over truncation in realizing digital filter?**

Error introduced due to rounding operation is less compared to truncation. Similarly quantization error due to rounding is independent of arithmetic operation and mean of rounding error is zero. Hence rounding is preferred over truncation in realizing digital filter.

6. **What is meant by finite word length effects in digital filters? (OR) Define finite word length effects**
   The digital implementation of the filter has finite accuracy. When numbers are represented in digital form, errors are introduced due to their finite accuracy. These errors generate finite precision effects or finite word length effects.
   When multiplication or addition is performed in digital filter, the result is to be represented by finite word length. Therfore the result is quantized so that it can be represented by finite word register. This quantization error can create noise or oscillations in the output. These effects are called finite word length effects.

7. **What is meant by dead band of the filter? (Or) Define dead band .**
   The limit cycle occurs as a result of quantization effect in multiplication. The amplitude of the output during a limit cycle are confined to a range of values called the dead of the fiter.
   Dead band= $\pm 2^{-B}/1\text{-}a$

8. **What are the methods used to prevent overflow?**
   There are two method used to prevent overflow: (i) saturation arithmetic (ii) sacling

9. **What are the different types in fixed point number representation?**
   (i)Sign-magnitude (ii) 1's complement (iii) 2's complement

10. **What are the three quantization errors due to finite word length registers in digital filters?**
    (i) Input quantization error,(ii) coefficient quantization error,(iii) product quantization error.

11. **How many multiplication and addition are carried out in floating point arithmetic?**
    In floating point arithmetic, multiplication are carried out as follows.
    Let $f_1=M_1x2^{c1}$ and $f_2= M_2x2^{c2}$. Then $f_3= f_1x\ f_2= (M_1x\ M_2)2^{(c_1+c_2)}$

That is mantissas are multiplied using fixed point arithmetic and the exponents are added.

The sum of two floating point numbers is carried out by shifting the bits of the mantissa of the smaller number to the right until the exponents of the two number are equal and then adding the mantissas.

12. **What do you mean by (zero-input) limit cycle oscillations?**
    When stable IIR filter digital filter is excited by finite sequence that is constant the output

will ideally decay to zero. However the non-linearity due to finite precision arithmetic operation often causes periodic oscillations to occur in the output. such oscillation occur in the recursive systems are called zero input limit cycle oscillation.

13. Define over flow error (or) overflow oscillations (or) over flow 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 error.

14. Define signal scaling.

Saturation arithmetic eliminates limit cycles due to overflow, but it causes undesirable signal distortion due to the nonlinearity of the clipper.

15. 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.

16. What are the advantages of floating point arithmetic?

(i) Larger dynamic range (ii) overflow in floating point representation is unlikely.

17. 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.

18. What do you understand by input quantization error?

In digital signal processing the continuous time input signals are converted into digital using a b-bit ADC. The representation of continuous signal amplitude by a fixed digit produces an error.

19. What are the different quantization methods?

(i) truncation (ii) Rounding

20. What is meant by rounding?

Rounding a number to b bits is accomplished by choosing the rounded result as the b bit number closest to the original number unrounded.

21. Why rounding is preferred to truncation in realizing digital filter?

(i) The mean of rounding error is zero

(ii) The variance of the rounding error signal is low.

22. What is meant by quantization step size?

Let us assume a sinusoidal signal varying between +1 and -1 having a dynamic range If ADC used to convert the sinusoidal signal employs b +1 bits including sign, the number levels available for quantizing $q=R/2^B$.

23. How overflow limit cycles can be eliminated?

The overflow limit cycles can be eliminated by two methods.

Saturation method: In this method if overflow occurs then output is set to maximum allowable value. And if underflow occurs then output is set to minimum allowable value. This introduces distortion in the output but overflow oscillations are eliminated.

Scaling: The input signaling to the address is properly scaled such that overflow is avoided.

24. What are the quantization errors due to finite wordlength registers in digital filters?

- Product round off errors.
- Limit cycle oscillations.

25. Compare truncation with rounding errors.

	Truncation error	Rounding error
1	Results from truncation of significant digits.	Results from rounding of significant digits.
2	As number of digits are more error is reduced	Rounding error depends upon value of digit being rounded.

26. What is meant by A/D conversion noise?

A digital signal processor contains a device, A/D converter that operates on the analog input x(t) to produce $x_q(n)$. The difference signal e(n) = xq(n)-x(n) is called A/D conversion noise.

27. What are the two kinds of limit cycle behavior in DSP?

- Zero input limit cycle oscillations.
- Overflow limit cycle oscillations.

28. What are the different types of arithmetic in digital systems?

There are three types of arithmetic used in digital system

- Fixed point arithmetic

- Floating point arithmetic
- Block Floating arithmetic

29. What do you understand by a fixed point number?

In fixed point arithmetic the position of the binary point is fixed. The bit to the right represent the fractional part of the number and those to the left represent the integer part.

30. What do you understand by 2's complement representation.?

In two's complement representation positive numbers are represented as in sign magnitude and one's complement. The negative number is obtained by complementing all the bits of the positive number and adding one to the least significant bit.

PART-B

1. Explain the characteristics of a limit cycle oscillation with respect to the system described by the difference equation y(n) =0.95y(n-1) + x(n). Determine the dead band of the filter.

2. The input to the system y(n) 0.999y(n-1) + x(n) is applied to an ADC. What is the power produced by the quantization noise at the output of the filter if the input is quantized to a (1) 8 bits (2) 16 bits.

3. The output signal of an A/D converter is passed through a first order low pass filter, with transfer function given by H(z) = (1-a)z / z-a for 0 <a< 1. Find the steady state output noise power due to quantization at the output of the digital filter.

4. Briefly explain the following:
   i) Coefficient quantization error
   ii) Product quantization error
   iii) Truncation and Rounding

5. Explain the quantization process and errors introduced due to quantization.

6. Explain th characteristics of limit cycle oscillation with respect to the system described by the difference equation:
   y(n) = 0.95 y(n-1) + x(n); x(n) =0 and y(-1) = 13.

7. Define zero input limit cycle oscillation and explain.

8. How is signal scaling used to prevent overflow limit cycle in the digital filter implementation? Explain with an example.

9. Study the limit cycle behavior of the system described by $y(n) = Q[ay(n-1)] + x(n)$, where y (n) is the output of the filter and Q[.] is quantization. Assume a= 7/8, x(0) = ¾ & x=0 , for n>0 choose 4 bit sign magnitude.

10. Find the effect of coefficient quantization on pole location of the given second order IIR system, when it is realized in direct form I and in cascade form. Assume a word length of 4 bits through truncation.

UNIT-V INTRODUCTION TO DIGITAL SIGNAL PROCESSORS

PART-A

1. What are the classification digital signal processors?

The digital signal processors are classified as
- General purpose digital signal processors
- 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. **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.

4. **What is pipelining? (OR) Define 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.

5. **What is pipeline depth?**

The number of pipeline stages is referred to as the pipeline depth.

6. What are the different stages in pipelining?(or) List the stage of pipelining.
- The fetch phase-An instruction is fetched from the memory.
- The decode phase-An instruction is decoded.
- Memory read phase-An operand required for the instruction is fetched from the data memory
- The execute phase-The operation is executed and results are stored at appropriate place.

7. What are different buses of TMS320c5x and their functions?

- Program bus
- Program address bus
- Data read bus
- Data read address bus

8. List the on chip peripherals in c5x.

Clock generator, Hardware timer, Software programmable wait state generators, General purpose I/O pins, Parallel I/O pins, Serial port interface, Buffered serial port, Time division multiplexed serial port, unmaskable interrupts.

9. Differentiate between Von Neumann and Harvard architectures.

In Von Neumann architectures the CPU can be either reading and instruction or reading/ writing data fromto the memory. Both cannot occur at the same time since the instruction and data use the same signal pathways and memory.

Whereas the Harvard architectures has two memories for their instructions and data requiring dedicated buses for each of them.

10. What is on-chip memory?

The memory present outside the processor is called the off-chip memory. If the memory is present within the processor block is called on-chip memory.

11. How to calculate the throughput of the pipeline?

The throughput of a pipeline machine is defined as the total number of instructions to be executed per unit time.

Throughput of the pipeline= Number of instruction/ Unit time.

12. What is barrel shifter?

Barrel shifters are used at the input and the output of the MAC unit. They do not require extra clock cycle to implement the shift operation.

13. What are the advantage and disadvantages of VLIW processor?

Advantages:

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

Disadvantages:

- High power consumption
- Misleading MIPS instruction

14. What are the comparison between RISC and CISC processor?

S.No	CISC	RISC
1	Emphasis on hardware	Emphasis on software
2	Includes multi-clock complex instructions.	Single clock, reduced instructions.
3	Small code size and high cycle per second	Large code size and low cycles per second
3	Transistors used for storing complex instructions	Spends more transistors on memory registers.

15. What are the features of TMS320C54 DSP processors?

Features of TMS320C54 DSP processors are,
16 bit CPU.
20 to 50 ns single cycle instruction execution time.
64K x 16 bit external program memory address space.
64K x 16 bit external data memory address space.
64K x 16 bit external IO address space.

16. What is meant by programmable digital signal processor?

A programmable digital signal processor has a general design configuration or architecture to cater to different applications by programming. It has more advantages over the advanced microprocessor and the RISC processor in terms of power requirement ,costs ,real time implementation and on chip memory.

17. What are the different kinds of architecture available for general purpose digital signal processor?

- Von-Neumann architecture
- Harvard architecture
- Modified architecture.

18. Define periodogram.

The periodogram a non-parametric method of power spectrum estimation .
Let the brased autocorrelation estimate $Y_{xx}(l)$
The Fourier transform of $Y_{xx}(l)$ is

19. What is MAC?

MAC- Multiplier accumulator is a hardware present in digital signal processor to perform multiplication and addition operation with less amount of time consumption.

20. What are the unique features of digital signal processor?

The unique features of digital signal processor is it has ADC , filtering, detection , enhancement, sampling, quantization and encoding functions.

21. What is the function of parallel logic unit?

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

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

Branch control input(BIO)

External flag (XF)

23. What are the shift instructions?

ROR, ROL, ROLB, RORB, BSAR.

24. What are the load/store instructions?

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

25. List the various registers used with ARAU.

Eight auxiliary registers. (AR0-AR7)

Auxiliary register pointer(ARP)

Unsigned 16-bit ALU.

26. What are the elements that the control processing unit of c5x consists.

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.

27. Mention the special features of DSP Processor.

- DSP processors should have multiple pointers to support multiple operands, jumps and shifts
- DSP processors should have multiple registers. So that data exchange from register to register is fast.
- DSP processors should have circular buffers to support circular shift operations.
- DSP processor should be able to perform multiply and accumulate operations very fast.

28. List the special features of Harvard architecture.

- The speed of execution in Harvard architecture is high. Because it have separate on chip memories and internal buses.
- The instruction code from program memory and operands from data memory can be fetched simultaneously. This parallel operation increases the speed.

29. Enumerate the addressing modes in DSP Processor.

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

30. List the elements in program controller of TMSC320C54?

The elements in program controller is given by, 16 bit program controller.

- 16 bit status register.
- Processor Mode Status Register (PMST).
- Circular Buffer Control Register (CBCR)
- Address Generation Logic.
- Instruction Register.

31. 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

32. Mention the instruction set in DSP Processor.

- Arithmetic instruction
- Logical instructionϖ Shift instruction
- Load / store instruction
- Move instruction
- Branch and call instruction
- Push and pop instruction
- RET Instruction
- Repeat Instruction

- In & out instruction
- List the commercial processors

33. What is BSAR instruction?

The contents of the accumulator are arithmetically right barrel shifted by 1 to 16 bits. This shift is defined in the shift operand of the instruction. This instruction is executed in single cycle. If SXM bit is cleared, the high order bits of the accumulator are zero filled. If SXM bit is set, the high order bits of accumulator are sign extended.

34. What is an Auxiliary Register Pointer (ARP)?

In indirect addressing mode, the data memory address is specified by the content of one of the eight Auxiliary Registers AR0 - AR7. The AR (Auxiliary Register) currently used for accessing data is denoted by ARP (Auxiliary Register Pointer).

35. Compare the functions of program bus and program address bus.

Program Bus	Program Address Bus
The program bus carries the instruction code and immediate operand from program memory to the CPU	The program address bus provides address to program memory space for both read and write

PART-B

1. (i) Draw the block diagram of Harvard architecture and explain.
   (ii)Explain the advantages and disadvantages of VLIW architecture.

2. Explain various addressing modes of a digital signal processor.

3. Draw the functional block diagram of a digital signal processor and explain.

4. Explain Von Neumann, Harvard architecture and modified Harvard architecture for the computer.

5. Explain about pipelining in DSP.

6. Explain the architecture of TMS320C50 with a neat diagram.

7. Describe the Architectural details and features of a DSP processor.