IT6502 DIGITAL SIGNAL PROCESSING

UNIT-I SIGNALS AND SYSTEMS

PART - A

1. What is a continuous and discrete time signal?

Continuous time signal: A signal x(t) is said to be continuous if it is defined for all time t. Continuous time signal arise naturally when a physical waveform such as acoustics wave or light wave is converted into a electrical signal.

Discrete time signal: A discrete time signal is defined only at discrete instants of time. The independent variable has discrete values only, which are uniformly spaced. A discrete time signal is often derived from the continuous time signal by sampling it at a uniform rate.

2. Give the classification of signals?

- Continuous-time and discrete time signals
- Even and odd signals
- Periodic signals and non-periodic signals
- Deterministic signal and Random signal
- Energy and Power signal

3.What are the types of systems?

- Continuous time and discrete time systems
- Linear and Non-linear systems
- Causal and Non-causal systems
- Static and Dynamic systems
- Time varying and time in-varying systems
- Distributive parameters and Lumped parameters systems
- Stable and Un-stable systems.

4.What are even and odd signals?

Even signal: continuous time signal x(t) is said to be even if it satisfies the condition x(t)=x(-t) for all values of t.

Odd signal: he signal x(t) is said to be odd if it satisfies the condition x(-t)=-x(t) for all t. In other words even signal is symmetric about the time origin or the vertical axis, but odd signals are anti-symmetric about the vertical axis.

5. What are deterministic and random signals?

Deterministic Signal: deterministic signal is a signal about which there is no certainty with respect to its value at any time. Accordingly, we find that deterministic signals may be modeled as completely specified functions of time.

Random signal: random signal is a signal about which there is uncertainty before its actual occurrence. (e.g.) The noise developed in a television or radio amplifier is an example for random signal.

6. What are energy and power signal?

Energy signal: Signal is referred as an energy signal, if and only if the total energy of the signal satisfies the condition $0<E<\infty$.

Power signal: Signal is said to be power signal if it satisfies the condition $0<P<\infty$.

7. What are elementary signals and name them?

The elementary signals serve as a building block for the construction of more complex signals. They are also important in their own right, in that they may be used to model many physical signals that occur in nature.

There are five elementary signals. They are as follows

1) Unit step function

2) Unit impulse function
3) Ramp function
4) Exponential function
5) Sinusoidal function

8. ***What are time invariant systems?***

A system is said to be **time invariant** system if a time delay or advance of the input signal leads to an identical shift in the output signal. This implies that a time invariant system responds identically no matter when the input signal is applied. It also satisfies the condition

$$T\{x\ (n-k)\} = y\ (n-k).$$

9. What do you mean by periodic and non-periodic signals?

A signal is said to be periodic if $x\ (n + N) = x\ (n)$, Where N is the time period.

A signal is said to be non-periodic if $x\ (n + N) = -x\ (n)$.

10. Define time variant and time invariant system.

A system is called **time invariant** if its output, input characteristics does not change with time. A system is called **time variant** if its input, output characteristics changes with time.

11. Define linear and non-linear system.

Linear system is one which satisfies superposition principle. Superposition principle: The response of a system to a weighted sum of signals be equal to the corresponding weighted sum of responses of system to each of individual input signal.

i.e., $T\ [a1x1(n)+a2x2(n)]=a1T[x1(n)]+a2T[x2(n)]$

A system, which does not satisfy superposition principle, is known as non-linear system.

12. Define causal and non-causal system.

The system is said to be **causal** if the output of the system at any time 'n' depends only on present and past inputs but does not depend on the future inputs. e.g.:- $y\ (n) =x\ (n)-x\ (n-1)$

A system is said to be **non-causal** if a system does not satisfy the above definition.

13. What are the steps involved in calculating convolution sum?

The steps involved in calculating sum are Folding, Shifting, Multiplication, Summation

14. Define causal LTI system.

The LTI system is said to be causal if $h\ (n) = 0$ for $n < 0$

15. Define stable LTI system.

The LTI system is said to be stable if its impulse response is absolutely summable.

16. What are the properties of convolution sum?

The properties of convolution sum are

- Commutative property
  The commutative law can be expressed as $x\ (n) * h\ (n)=h\ (n) * x\ (n)$
- Associative law
  The associative law can be expressed as $[x\ (n) * h1\ (n)] * h2\ (n) = x\ (n)\ [h1\ (n) * h2\ (n)]$ Where x (n) – input, h1 (n), h2 (n) - impulse response
- Distributive law
  The distributive law can be expressed as
  $x\ (n) * [h1\ (n) + h2\ (n)] = x\ (n) * h1\ (n) + x\ (n) * h2\ (n)$

17. Define Region of convergence

The region of convergence (ROC) of X (Z) is the set of all values of Z for which X (Z) attain final value.

18 .State properties of ROC

x The ROC does not contain any poles.
H When x (n) is of finite duration then ROC is entire Z-plane except $Z = 0$ or $Z = \infty$
I If X (Z) is causal, then ROC includes $Z = \infty$

J If X (Z) is non-casual, then ROC includes Z = 0

19 . Continuous time and Discrete time signals.

Continuous Time (CTS)	**Discrete time (DTS)**
This signal can be defined at any time instance & they can take all values in the continuous interval(a, b) where a can be ∞ & b can be ∞	This signal can be defined only at certain specific values of time. These time instance need not be equidistant but in practice they are usually takes at equally spaced intervals.
These are described by differential equations.	These are described by difference equation.
This signal is denoted by x(t).	These signals are denoted by x(n) or

21. Analog and digital signal.

Analog signal	**Digital signal**
They are basically continuous time & continuous amplitude signals.	They are basically discrete time signals & discrete amplitude signals. These signals are basically obtained by sampling & quantization process.
ECG signals, Speech signal, Television signal etc. All the signals generated from various sources in nature are analog.	All signal representation in computers and digital signal processors are digital.

PART – B (16 Marks)

1. Explain in detail about the classification of discrete time systems. (16)

2.(a) Describe the different types of discrete time signal representation. (6)

(b) Define energy and power signals. Determine whether a discrete time unit step signal x(n) = u(n) is an energy signal or a power signal. (10)

3.(a) Give the various representation of the given discrete time signal x(n) = {-1,2,1,-2,3} in Graphical, Tabular, Sequence, Functional and Shifted functional. (10)

(b) Give the classification of signals and explain it. (6)

4.(a) Draw and explain the following sequences:

i) Unit sample sequence ii) Unit step sequence iii) Unit ramp sequence

iv) Sinusoidal sequence and v) Real exponential sequence (10)

5..Determine if the system described by the following equations are causal or noncausal

i)y (n) = x(n) + (1 / (x(n-1))

ii) y(n) = x(n2) (6)

6. Determine the values of power and energy of the following signals. Find whether the signals are power, energy or neither energy nor power signals.

x(n) = (1/3)n u(n)

x(n) = e j((π/2)n + (π/4)

x(n) = sin (π/4)n

x(n) = e2n u(n) (16)

7. Determine if the following systems are time-invariant or time-variant

y(n) = x(n) + x(n-1)

y(n) = x(-n) (16)

8. Determine if the system described by the following input-output equations are linear or non-linear.

1. y(n) = x(n) + (1 / (x(n-1))
2. y(n) = x2(n)

3. y(n) = nx(n) (16)

9. Test if the following systems are stable or not.
   1. y(n) = cos x(n)
   2. y(n) = ax(n)
   3. y(n) = x(n) en
   4. y(n) = ax(n) (16)

10. (a) Explain the principle of operation of analog to digital conversion with a neat diagram. (8)
(b) Explain the significance of Nyquist rate and aliasing during the sampling of continuous time signals. (8)

11.. (a) List the merits and demerits of Digital signal processing. (8)
(b) Write short notes about the applications of DSP. (8)

12.(a) Find the convolution of the following sequences
i) x(n)=u(n) h(n)=u(n-3) ii) x(n)={1,2,-1,1} h(n)={1,0,1,1} (8)
(b) Determine the response of the causal system y(n)-y(n-1)=x(n)+x(n-1) to inputs x(n)=u(n) and x(n)=2 –n u(n). (8)

13. (a) Determine the solution of the difference equation
y(n) = 5/6 y(n-1) – 1/6 y(n-2) + x(n) for x(n) = 2n u(n) (8)
b)Determine the response y(n), n>=0 of the system described by the second order difference equation y(n) – 4y(n-1) + 4y(n-2) = x(n) – x(n-1) when the input is x(n) = (-1)n u(n) and the initial condition are y(-1) = y(-2)=1.

14. State and prove any two properties of z-transform. (8)

15. Find the z-transform and ROC of the causal sequence. X(n) = {1,0,3,-1,2} (8)

16. Find the z-transform and ROC of the anticausal sequence X(n) = {-3,-2,-1,0,1} (8)

17. (a) Determine the z-transform and ROC of the signal x(n) = anu(n)
(b) Find the stability of the system whose impulse response h(n) = (2)nu(n) (16)

18. (a) Determine the z-transform of x(n) = cos ωn u(n) (8)
(b) State and prove the following properties of z-transform. i) Time shifting ii) Time reversal iii) Differentiation iv) Scaling in z-domain (8)

19. Determine the inverse z-transform of x(z) = (1+3z-1) / (1+3z-1+2z-2) for z >2 (8)

20. Find the inverse z-transform of x(z) = (z2+z) / (z-1)(z-3), ROC: z > 3.
Using (i) Partial fraction method,
(ii) Residue method and
(iii) Convolution method (16)

21. Determine the unit step response of the system whose difference equation is
y(n)-0.7y(n-1)+0.12y(n-2) = x(n-1)+x(n-2) if y(-1) = y(-2) = 1. (8)

22. a) Determine the convolution sum of two sequences x(n) = {3,2,1,2}, h(n) = {1,2,1,2} (8)
b) Find the convolution of the signals x(n) = 1 n = -2,0,1 = 2 n = -1 = 0 elsewhere h(n) = δ(n)-δ(n-1)+ δ(n-2)- δ(n-3) (8)

UNIT-II DISCRETE FOURIER TRANSFORM

PART – A

1. State the properties of DFT?

1) Periodicity
2) Linearity and symmetry
3) Multiplication of two DFTs
4) Circular convolution
5) Time reversal
6) Circular time shift and frequency shift

7) Complex conjugate
8) Circular correlation

2. Define circular convolution?

Let x1(n) and x2(n) are finite duration sequences both of length N with DFTs X1(K) and X2(k). If X3(k)=X1(k)X2(k) then the sequence x3(n) can be obtained by circular convolution defined as

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

3. How to obtain the output sequence of linear convolution through circular convolution?

Consider two finite duration sequences x(n) and h(n) of duration L samples and M samples. The linear convolution of these two sequences produces an output sequence of durationL+M-1 samples, whereas, the circular convolution of x(n) and h(n) give N samples where N=max(L,M).In order to obtain the number of samples in circular convolution equal to L+M-1,both x(n) and h(n) must be appended with appropriate number of zero valued samples. In other words by increasing the length of the sequences x (n) and h(n) to L+M -1 points and then circularly convolving the resulting sequences we obtain the same result as that of linear convolution.

4. Define sectional convolution.

If the data sequence x(n) is of long duration it is very difficult to obtain the output sequence y(n) due to limited memory of a digital computer. Therefore, the data sequence is divided up into smaller sections. These sections are processed separately one at a time and controlled later to get the output.

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

No.	Linear convolution	Circular convolution
1	If x(n) is a sequence of L number of samples and h(n) with M number of samples, after convolution y(n) will have N=L+M-1 samples.	If x(n) is a sequence of L number of samples and h(n) with M samples, after convolution y(n) will have N=max(L,M) samples.
2	It can be used to find the response of a linear filter.	It cannot be used to find the response of a 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 filter.

6. What are differences between overlap-save and overlap-add methods.

No	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 L new data points	Each data block is L points and we append M-1 zeros to compute N point DFT
3	In each output block M-1 points are corrupted due to aliasing as circular convolution is employed	In this no corruption due to aliasing as linear convolution is performed using circular convolution
4	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

7. What the differences and similarities between DIF and DIT algorithms?

Differences:

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

Similarities:

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

8. 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 zero padding are
1) We can get better display of the frequency spectrum.
2) With zero padding the DFT can be used in linear filtering.

9. What is overlap-add method?

In this method the size of the input data block x i(n) is L. To each data block we append M-1 zeros and perform N point circular convolution of xi (n) and h(n). Since each data block is terminated with M-1 zeros the last M-1 points from each output block must be overlapped and added to first M-1 points of the succeeding blocks. This method is called overlap-add method.

10. What is overlap-save method?

In this method the data sequence is divided into N point sections xi(n). Each section contains the last M-1 data points of the previous section followed by L new data points to form a data sequence of length N=L+M-1.In circular convolution of xi(n) with h(n) the first M-1 points will not agree with the linear convolution of xi(n) and h(n) because of aliasing, the remaining points will agree with linear convolution. Hence we discard the first (M-1) points of filtered section x i (n) N h(n). This process is repeated for, all sections and the filtered sections are abutted together.

11. Why FFT is needed?

The direct evaluation DFT requires N2 complex multiplications and N2 –N complex additions. Thus for large values of N direct evaluation of the DFT is difficult. By using FFT algorithm the number of complex computations can be reduced. So we use FFT.

12. 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 to effectively reduce the DFTcomputation time. It is based on the fundamental principle of decomposing the computation of DFT of a sequence of length N into successively smaller DFTs.

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

The number of multiplications and additions required to compute N point DFT using radix-2 FFT are N log2 N and N/2 log2 N respectively.

14. 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 algorithm.

15. What is DIT algorithm?

Decimation-In-Time algorithm is used to calculate the DFT of a N point sequence. The idea is to break the N point sequence into two sequences, the DFTs of which can be combined to give the DFT of the original N point sequence. This algorithm is called DIT because the sequence x(n) is often splitted into smaller sub- sequences.

16. What DIF algorithm?

It is a popular form of the FFT algorithm. In this the output sequence X(k) is divided into smaller and smaller sub-sequences, that is why the name Decimation In Frequency (DIF).

17. What are the applications of FFT algorithm? The applications of FFT algorithm includes

1) Linear filtering
2) Correlation
3) Spectrum analysis

18. What is the relationship between z-transform and DTFT? The

z-transform of x(n) is given by

$$x(z) = \sum_{n=-\infty}^{\infty} x(n)\, Z^{-n} \quad ; \text{where } z = re^{j\omega} \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ (1)$$

Substituting z in x(z) we get,

$$x(z) = \sum_{n=-\infty}^{\infty} x(n)\, r^{-n} e^{-j\omega n} \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ (2)$$

The Fourier transform of x(n) is given by

$$x(e^{j\omega}) = \sum_{n=-\infty}^{\infty} x(n)\, e^{-j\omega n} \qquad \ldots\ldots\ldots\ldots\ldots\ldots(3)$$

Equation (2) and (3) are identical, when r = 1.

In the z-plane this corresponds to the locus of points on the unit circle |Z|=1.
Hence $X(e^{jw})$ is equal to H(z) evaluated along the unit circle, or $X(e^{jw}) = x(z)|_{z=e^{jw}}$
For $X(e^{jw})$ to exist, the ROC of x(z) must include the unit circle.

19. 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 x(k). The N point DFT of a finite sequence x(n) of length L,(L<N) is defined as

$$x(k)=\sum_{n=0}^{N-1} x(n)e^{-j2\pi nk/N} \qquad K=0,1,2,3,\dots N-1$$

20. Define IDFT?

The IDTFT of the sequence of length N is defined as

$$X(n)=(1/N)\sum_{k=0}^{N-1} x(k)e^{j2\pi nk/N} \qquad n=0,1,2,3,\dots N-1$$

21. Define DTFT and IDTFT of a sequence?

The DTFT (Discrete Time Fourier Transform) of a sequence x(n) is defined as

$$X(w) = \sum_{n=-\infty}^{\infty} x(n)e^{-jwn}$$

The IDTFT is defined as

$$x(n)= 1/2\pi \int_{-\pi}^{\pi} x(w)\, e^{jwn}\, dw$$

22. What is the drawback in DTFT?

The drawback in discrete time fourier transform is that it is continuous function of w and cannot be processed by digital systems.

23. State periodicity property with respect to 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.

24. State linearity property with respect to DFT?

If x1(k) and x2(k) are N-point DFTs of finite duration sequences x1(n) and x2(n), then DFT [a x1(n) + b x2(n)] = a x1(k) + b x2(k), a, b are constants.

25. State time reversal property with respect to DFT?

If DFT[x(n) =x(k), then
DFT[x((-n))N] = DFT[x(N-n)] = x((-k))N = x(N-k)

26. State circular time shifting property with respect to DFT? If

DFT[x(n)] = x(k), then DFT [x((n-l))] = x(k) $e^{-j2\Pi k1/N}$

27. What is the basic operation of DIF algorithm? (University)

The basic operation DIF algorithm is called butterfly in which two inputs G(n) and H(n)are combined to give x1(k) and x2(k)

$$x_1(k)= G(n)+ H(n)$$

$$x_2(k)=\{G(n)- H(n)\}\, W_N^k$$

Where, W_N^k is the **twiddle factor**

PART B – 16 Marks

1. (a) Determine the output response y(n) if h(n) = {1,1,1,1}; x(n) = {1,2,3,1} by using (i) Linear convolution ii) Circular convolution and iii) Circular convolution with zero padding (12)

(b) Explain any two properties of Discrete Fourier Transform. (4)

2. Using linear convolution find y (n) = x (n)*h (n) for the sequences x (n) = (1,2,-1,2,3,-2,-3,-1,1,1,2,-1) and h(n) = (1,2).Compare the result by solving the problem using i) Over-lap save method and ii) Overlap – add method.

3. Describe the decimation in time [DIT] radix-2 FFT algorithm to determine N-point DFT.

4. An 8-point discrete time sequence is given by x(n) = {2,2,2,2,1,1,1,1}. Compute the 8-point DFT of x (n) using radix-2 FFT algorithm. (16)

5. (a) Compute the 4-point DFT and FFT-DIT for the sequence x(n) = {1,1,1,3} and What are the basic steps for 8-point FFT-DIT algorithm computation? (12)

b)What is the advantage of radix-2 FFT algorithm in comparison with the classical DFT method?

6. (a) Perform circular convolution of the two sequences graphically x1(n) = {2,1,2,1} and x2(n)= {1,2,3,4} (6)

(b) Find the DFT of a sequence by x(n) = {1,2,3,4,4,3,21} using DIT algorithm. (10)

7. (a) Explain the decimation in frequency radix-2 FFT algorithm for evaluating N-point DFT of the given sequence. Draw the signal flow graph for N=8. (12)

(b) Find the IDFT of y (k) = {1,0,1,0} (4)

8. (a) Find the circular convolution of the sequences x1(n)= {1,2,3} and x2(n) = {4,3,6,1} (8)

(b) Write the properties of DFT and explain. (8)

9. (a) Draw the 8-point flow diagram of radix-2 DIF-FFT algorithm. (8)

(b) Find the DFT of the sequence x(n) = {2,3,4,5} using the above algorithm. (8)

10. (a) What are the differences and similarities between DIT and DIF FFT algorithms? (6)

(b) Compute the 8-point 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 DIT algorithm. (10)

11. (a) Compute the 8-point DFT of the sequence x (n) = {0.5,0.5,0.5,0.5,0,0,0,0} using radix-2 DIT algorithm. (8)

(b) Find the IDFT of the sequence x (k) = {4,1-j2.414,0,1-j0.414,0,1+j.414,0,1+j2.414} using DIF algorithm. (8)

12. Compute the 8-point DFT of the sequence x(n) = 1, $0 \le n \le 7$
0, otherwise

by using DIT,DIF algorithms. (16)

UNIT-III IIR FILTER DESIGN

PART – A

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

Based on impulse response the filters are of two types

1) IIR filter
2) FIR filter

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

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

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

Based on frequency response the filters can be classified as

i. Low pass filter
ii. High pass filter
iii. Band pass filter
iv. Band reject filter

3. What are the advantages and disadvantages of FIR filters?

Advantages:

c. FIR filters have exact linear phase.
d. FIR filters are always stable.
e. FIR filters can be realized in both recursive and non recursive structure.
f. Filters with any arbitrary magnitude response can be tackled using FIR sequence.

Disadvantages:

6) For the same filter specifications the order of FIR filter design can be as high as 5 to 10 times that in an IIR design.
7) Large storage is needed.
8) Powerful computational facilities required for the implementation.

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

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.

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

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

- Impulse invariance method.
- Bilinear transformation method.
- Approximation of derivatives

6. Distinguish between FIR filters and IIR filters.

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

7. What do you understand by backward difference?

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

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

The above equation is called backward difference equation.

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

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

In this method of digitizing an analog filter, the impulse response of the resulting digital filter is a sampled version of the impulse response of the analog filter. If the transfer function is of the form, 1/s-p, then

$$K(z) = 1/1-e-pTz-1$$

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 is in bilinear transformation is

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

11. What are the properties of bilinear transformation?

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

12. Write a short note on pre-warping.

The effect 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 pre-scaling, or pre-warping the critical frequencies by using the formula.

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

Advantages:

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

Disadvantage:

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

14. Distinguish analog and digital filters

Analog Filter	**Digital Filter**

Constructed using active or passive components and it is described by a differential equation	Consists of elements like adder, subtractor and delay units and it is described by a difference equation
Frequency response can be changed by changing the components	Frequency response can be changed by changing the filter coefficients
It processes and generates analog output	Processes and generates digital output
Output varies due to external conditions	Not influenced by external conditions

15. What are the properties of chebyshev filter?

- The magnitude response of the chebyshev filter exhibits ripple either in the stop band or the pass band.
- The poles of this filter lies on the ellipse

16. List the Butterworth polynomial for various orders.

N	Denominator polynomial
1	S+1
2	S^2+.707s+1
3	(s+1) (s2+s+1)
4	(s^2+.7653s+1) (s^2+1.84s+1)
5	(s+1) (s^2+.6183s+1) (s2+1.618s+1)
6	(s^2+1.93s+1) (s^2+.707s+1) (s^2+.5s+1)
7	(s+1) (s^2+1.809s+1) (s^2+1.24s+1) (s^2+.48s+1)

17. Differentiate Butterworth and Chebyshev filter.

Butterworth damping factor 1.44 chebyshev 1.06 Butterworth flat response damped response.

18. Which types of structures are used to realize IIR systems?

* Direct form -1structure (DF-1)
* Direct form -2structure (DF-2)
* Cascade form structure
* Parallel form structure

19. Why direct form-II structure is preferred most and why?

The numbers of delay elements are reduced in direct form-II structure compared to direct form-I structure. That means the memory locations are reduced in direct form-II structure.

20. Distinguish direct-I and direct-II forms.

The **direct-form I** realization requires M+N+1 multiplications, M+N additions and M+N+1 memory locations.

The **direct-form II** realization requires M+N+1 multiplications, M+N additions and the maximun of (M,N) memory locations.

21. What is warping effect or frequency warping?

The relation between the analog and digital frequencies in bilinear transformation is given by, Ω = tan . For smaller values of ω, there exists linear relationship between ω and Ω .But for Larger values of ω, the relationship is nonlinear. This introduces distortion in the frequency axis. This effect compresses the magnitude and phase response. This Effect is called warping effect.

PART – B

1. a) Derive bilinear transformation for an analog filter with system function H(S) = b/S + a
   b) Discuss the limitation of designing an IIR filter using impulse invariant method
2. Determine H (Z) for a Butterworth filter satisfying the following specifications:

$$0.8 \le |H(e^{j\omega})| \le 1, \text{ for } 0 \le \omega \le \pi/4$$
$$|H(e^{j\omega})| \le 0.2, \text{ for } \pi/2 \le \omega \le \pi$$

   Assume T= 0.1 sec. Apply bilinear transformation method
3. Determine digital Butterworth filter satisfying the following specifications:

$$0.707 \le |H(e^{j\omega})| \le 1, \text{ for } 0 \le \omega \le \pi/2$$
$$|H(e^{j\omega})| \le 0.2, \text{ for } 3\pi/4 \le \omega \le \pi$$

Assume T= 1 sec. Apply bilinear transformation method.

4. Design a Chebyshev low pass filter with the specifications $\alpha_p=1$ dB ripple in the pass band $0\le\omega\le0.2\pi$, $\alpha_s=15$ dB ripple in the stop band $0.3\pi\le\omega\le\pi$ using impulse invariance method
   $\alpha_p=1$ dB; $\alpha_s=15$ dB; $\Omega_p=0$
5. Design a Butterworth high pass filter satisfying the following specifications.
6. Design a Butterworth low pass filter satisfying the following specifications.

$$f_p=0.10 \text{ Hz}: \alpha_p=0.5 \text{ dB}$$
$$f_s=0.15 \text{ HZ}: \alpha_s=15 \text{ dB}: F=1\text{Hz}.$$

7. Design (a) a Butterworth and (b) a Chebyshev analog high pass filter that will pass all radian frequencies greater than 200 rad/sec with no more that 2 dB attuenuation and have a stopband attenuation of greater than 20 dB for all Ω less than 100 rad/sec.
8. Design a digital filter equivalent to this using impulse invariant method H(S) = 10/S2+7S+10
9. Use impulse invariance to obtain H(Z) if T= 1 sec and H(s) is

$$1/(s^3+3s^2+4s+1)$$
$$1/(s^2+\sqrt{2}\,s+1)$$

10. Use bilinear transformation method to obtain H (Z) if T= 1 sec and H(s) is

$$1/(s+1)(S+2)$$
$$1/(s^2+\sqrt{2}\,s+1)$$

11. Briefly explain about bilinear transformation of digital filter design
12. Compare bilinear transformation and impulse invariant mapping
13. Design a chebyshev filter with a maxmimum pass band attenuation of 2.5 Db; at Ωp=20 rad/sec and the stop band attenuation of 30 Db at Ωs=50 rad/sec.
14. Describe various Structures of IIR filter .

UNIT- IV FIR FILTER DESIGN

PART – A

1. How phase distortion and delay distortion are introduced?

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

2. What is mean by FIR filter?

The filter designed by selecting finite number of samples of impulse response h (n) obtained from inverse Fourier transform of desired frequency response H(w) are called FIR filters.

3.Write the steps involved in FIR filter design

- Choose the desired frequency response Hd(w)
- Take the inverse fourier transform and obtain hd(n) and convert the infinite duration sequence Hd(n) to h(n) . Take z transform of h(n) to get H(z)

4. Give the advantages of FIR filter?

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

5. List the disadvantages of FIR FILTER

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

6. Define necessary and sufficient condition for the linear phase characteristic of a FIR filter?

The phase function should be a linear function of w, which in turn requires constant group delay and phase delay.

8. List the well-known design technique for linear phase FIR filter design?

Fourier series method and window method

Frequency sampling method

9. For what kind of application, the anti-symmetrical impulse response can be used?

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

9. 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., lowpass, highpass, bandpass and band reject. The symmetric impulse response having even number of samples can be used to design lowpass and bandpass filter.

10. Justify that that FIR filter is always stable?

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

11. 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

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

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

If the impulse response is anti symmetrical, 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.

13. 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.

14. When cascade from realization is preferred in FIR filters?

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

15. What are the disadvantages of Fourier series method?

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

16. Define Gibbs phenomenon? OR What are Gibbs oscillations?

One possible way of finding an FIR filter that approximates $H(e^{jw})$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 is stop band .This phenomenon is known as Gibbs phenomenon.

17. Give the desirable characteristics of the windows?

- The desirable characteristics of the window are
- The central lobe of the frequency response of the window should contain most of the energy and should be narrow.
- The highest side lobe level of the frequency response should be small.
- The side lobes of the frequency response should decrease in energy rapidly as w tends to p.

18. What is the necessary and sufficient condition for linear phase characteristics in FIR filter?

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

19. What are the advantages of Kaiser Widow?

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

20. What is the principle of designing FIR filter using frequency sampling method?

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

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

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

22. State Frequency Warping

Because of the non-linear mapping: the amplitude response of digital IIR filter is expand at lower frequencies and compressed at higher frequencies in comparison to the analog filter.

23. What is the importance of poles in filter design?

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

24. What are the applications of FIR filter?

Symmetric Response

To design all types of filter such as HPF, LPF, BPF, BSF

Antisymmetric Response

To design Hilbert Transformer and Differentiator

PART – B (16 – MARKS)

1. Explain the design of FIR filters using frequency sampling method.

2. State and explain the properties of FIR filters. State their importance. (8)
3. Explain linear phase FIR structures. What are the advantages of such structures?
4. Write the expressions for the Hamming, Hanning and Rectangular windows.
5. Explain the design of FIR filters using windows.
6. Design an ideal Low pass filter with a frequency response

$H_d(e^{jw}) = 1$ for $\pi/4 \le |w| \le \pi$
$= 0$ for $|w| \le \pi/4$

Find the values of h (n) for N = 11 using Rectangular window. Find H (z) and determine the magnitude response.
7. Design an ideal high pass filter with a frequency response

$H_d(e^{jw}) = 1$ for $\pi/4 \le |w| \le \pi$
$= 0$ for $|w| \le \pi/4$

Find the values of h (n) for N = 11 using hamming window. Find H (z) and determine the magnitude response.
8.Design an ideal Band pass filter with a frequency response

$H_d(e^{jw}) = 1$ for $\pi/4 \le |w| \le \pi$
$= 0$ for $|w| \le \pi/4$

Find the values of h (n) for N = 11 using hanning window. Find H (z) and determine the magnitude response.
9.Design an ideal Band pass filter with a frequency response

$H_d(e^{jw}) = 1$ for $\pi/4 \le |w| \le \pi$
$= 0$ for $|w| \le \pi/4$

Find the values of h (n) for N = 11 using Recatnagular window. Find H (z) and determine the magnitude response.
10. Obtain the direct form I, direct form II and Cascade form realization of the following system functions.
Y (n) = 0.1 y (n-1) + 0.2 y(n-2) + 3x(n) + 3.6 x(n-1) + 0.6 x(n-2).

UNIT V FINITE WORDLENGTH EFFECTS

PART – A

1. Define 1's complement form?
In 1,s complement form the positive number is represented as in the sign magnitude form. To obtain the negative of the positive number, complement all the bits of the positive number.
2. What is meant by 2's complement form?
In 2's complement form the positive number is represented as in the sign magnitude form. To obtain the negative of the positive number, complement all the bits of the positive number and add 1 to the LSB.
3. **Define floating pint representation?**
In floating point form the positive number is represented as F =2CM,where is mantissa, is a fraction such that1/2<M<1and C the exponent can be either positive or negative.
4.List the advantages of floating pint representation?

1.Large dynamic range
2. Overflow is unlikely.

5. Give the different quantization errors occurdue to finite word length registers in digital filters?

1. Input quantization errors

2. Coefficient quantization errors
3. Product quantization errors

6. What do you understand by input quantization error?

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

7 .Define product quantization error?

The product quantization errors arise at the output of the 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 will be rounded or truncated to b bits which produce the error.

8. Mention the different quantization methods available for Finite Word Length Effects?

Truncation
Rounding

9. State truncation?

Truncation is a process of discarding all bits less significant than LSB that is retained

10. Define Rounding?

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

11. List the two types of limit cycle behavior of DSP?

- Zero limit cycle behavior
- Over flow limit cycle behavior

12. Mention the methods to prevent overflow?

- Saturation arithmetic and
- Scaling

13. Give the different types of arithmetic in digital systems.

There are three types of arithmetic used in digital systems. They are fixed point arithmetic, floating point, block floating point arithmetic.

14. What is meant by fixed point number?

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

15.What are the different types of fixed point arithmetic?

Depending on the negative numbers representation, there are three forms of fixed point arithmetic. They are sign magnitude, 1's complement, 2's complement

16.Distinguish between fixed point and floating point arithmetic

S.No	Fixed Point Arithmetic	Floating Point Arithmetic
1.	Fast Operation	Slow Operation
2.	Relatively Economical	More expensive because of costlier hardware
3.	Overflow occurs in addition	Overflow does not arise
4.	Used in small computers	Used in large general purpose Computers
5.	Small Dynamic Range	Increased dynamic range

17. What do you understand by (Zero input) Limit cycle oscillations?

When a stable IIR filter digital filter is excited by a finite sequence, that is constant, the output will ideally decay to zero. However, the non-linearity due to finite precision arithmetic operations often causes periodic oscillations to occur in the output. Such oscillations occur in the recursive systems are called Zero input Limit Cycle Oscillation.

18. Determine Dead Band of the Filter.

The Limit cycle occurs as a result of quantization effect in multiplication. The amplitude of output during a limit cycle are confined to a range of values called the dead band of the filter.

19. Why Rounding is preferred to truncation in realizing digital filter?

- The quantization error due to rounding is independent of type arithmetic
- The mean of rounding error is zero
- The variance of rounding error is low

20. Define overflow oscillations

The overflow caused by adder makes the filter output to oscillate between maximum amplitude limits and such oscillations is referred as overflow oscillations

21. What is meant by sign magnitude representation?

For sign magnitude representation the leading binary digit is used to represent the sign. If it is equal to 1 the number is negative, otherwise it is positive.

PART – B (16 – MARKS)

1. Explain the limit cycle oscillations due to product round off and overflow errors?
2. Discuss in detail the errors resulting from rounding and truncation?
3. Explain how reduction of product round-off error is achieved in digital filters?
4. Explain the effects of co-efficient quantization in FIR filters?
5. Distinguish between fixed point and floating point arithmetic
6. With respect to finite word length effects in digital filters, with examples discuss about Over flow limit cycle oscillation and Signal scaling
7. What is called quantization noise? Derive the expression for quantization noise power.
8. Compare the truncation and rounding errors using fixed point and floating point representation.
9. Represent the following numbers in floating point format with five bits for mantissa and three bits for exponent.
   i. 710
   ii. 0.2510
   iii. -710
   iv. -0.2510
10. Determine the dead band of the system $y(n) = 0.2y(n-1) + 0.5y(n-2) + x(n)$ Assume 8 bits are used for signal representation.