SYED AMMAL ENGINEERING COLLEGE
(Approved by the AICTE, New Delhi, Govt. of Tamilnadu and Affiliated to Anna University, Chennai)
Established in 1998 - An ISO 9001:2008 Certified Institution
Dr. E.M.Abdullah Campus,Ramanathapuram – 623 502.
Phone: 304001, 304002 (04567) Fax: 304123(04567
Department of Electronics and Communication Engineering

IT6502/DIGIGITAL SIGNAL PROCESSING
Two MARKS Questionwith Answers

UNIT-I/ SIGNALS AND SYSTEMS

1. Define Signal.

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

2. Define a system.

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

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

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

4. Give some applications of DSP?

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

5. Write the classifications of DT Signals.

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

6. What is an Energy and Power signal?

Energy signal: A finite energy signal is periodic sequence, which has a finite energy but zero average power.

Power signal: An Infinite energy signal with finite average power is called a power signal.

7. What is Discrete Time Systems?

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

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

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

9. Define Linear system

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

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

10. Define Static & Dynamic systems

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

11. Define causal system.

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

12. Define Shift-Invariant system.

If y (n) is the response to an input x(n), then the response to an input X(n) = x(n-n0) then y(n) = y(n-n0) When the system satisfies above condition then it is said to shift in variant, otherwise it is variant.

13. Define impulse and unit step signal.

Impulse signal δ(n): The impulse signal is defined as a signal having unit magnitude at n = 0 and zero for other values of n. δ (n) = 1; n = 0
0; n ≤ 0

Unit step signal u(n): The unit step signal is defined as a signal having unit magnitude for all values of n. u(n) = 1; n ≥ 0
0; n≤ 0

14. What are FIR and IIR systems?

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

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

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

16. What is ROC in Z-Transform?

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

17. List any four properties of Z-Transform.

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

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

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

19. Define sampling theorem.

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

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

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

21. What are the properties of convolution?

1. Commutative property x(n) * h(n) = h(n) * x(n)
2. Associative property [x(n) * h1(n)]*h2(n) = x(n)*[h1(n) * h2(n)]
3. Distributive property x(n) *[h1(n)+h2(n)] = [x(n)*h1(n)]+[x(n) * h2(n)]

22. Define Z transform.

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

23. Define ROC.

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

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

x(n)= {1,2,3,4}

X(z)= x(n)z^{-n}

= 1+2z-1+3z-2+4z-3.

= 1+2/z+3/z2+4/z3.

25. State the convolution property of Z transform.

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

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

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

27. State initial value theorem.

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

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

Given X(z)=z-1/1-az-1

By time shifting property X(n)=an.u(n-1)

UNIT-II / FREQUENCY TRANSFORMATIONS

1. Define DTFT.

Let us consider the discrete time signal x(n).Its DTFT is denoted as X(z).It is given as

$$\mathbf{X(z)= x(n)e^{-jwn}}$$

2. State the condition for existence of DTFT?

The conditions are

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

3. List the properties of DTFT.

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

4. What is the DTFT of unit sample?

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

5. Define DFT.

DFT is defined as $\mathbf{X(z)= x(n)e^{-jwn}}$. Here x(n) is the discrete time sequence X(z) is the fourier transform of x(n).

6. Define Twiddle factor.

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

7. Define Zero padding.

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

8. Define circularly even sequence.

A Sequence is said to be circularly even if it is symmetric about the point zero on the circle.

$$\mathbf{x(N-n)=x(n),1<=n<=N-1.}$$

9. Define circularly odd sequence.

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

10. Define circularly folded sequences.

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

11. State circular convolution.

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

12. State parseval's theorem.

Consider the complex valued sequences x(n) and y(n).

If x(n)y*(n)=1/N X(k)Y*(k)

13. What is DFT?

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

14. Define N point DFT.

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

15. What is DFT of unit impulse δ(n)?

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

16. List the properties of DFT.

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

17. State Linearity property of DFT.

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

18. When a sequence is called circularly even?

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

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

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

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

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

$$N=L+M-1$$

Where

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

21. What is circular time shift of sequence?

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

22. What is the disadvantage of direct computation of DFT?

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

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

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

1. Radix-2 FFT algorithms. -Radix-2 Decimation in Time (DIT) algorithm.
   - Radix-2 Decimation in Frequency (DIF) algorithm.
2. Radix-4 FFT algorithm.

24. What is the computational complexity using FFT algorithm?

1. Complex multiplications = N/2 log2N
2. Complex additions = N log2N

25. How linear filtering is done using FFT?

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

26. 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 sequencex (n). This is known as zero padding. The uses of padding a sequence with zeros are

(i) We can get 'better display' of the frequency spectrum.

(ii) With zero padding, the DFT can be used in linear filtering.

27. Why FFT is needed?

The direct evaluation of the DFT using the formula requires N2 complex multiplications and N (N-1) complex additions. Thus for reasonably large values of N (inorder of 1000) direct evaluation of the DFT requires an inordinate amount of computation. By using FFT algorithms the number of computations can be reduced. For example, for an N-point DFT, The number of complex multiplications required using FFT is N/2log2N. If N=16, the number of complex multiplications required for direct evaluation of DFT is 256, whereas using DFT only 32 multiplications are required.

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

The number of complex multiplications required using direct computation is N2=642=4096. The number of complex multiplications required using FFT is
N/2 log2N = 64/2log264=192.Speed improvement factor = 4096/192=21.33

29. What is the main advantage of FFT?

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

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

For N-point DFT the number of complex multiplications needed using FFT algorithm is N/2 log2N. For N=32, the number of the complex multiplications is equal to 32/2log232=16*5=80.

31. What is FFT?

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

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

The number of multiplications and additions required to compute N-point DFT using redix-2 FFT are N log2N and N/2 log2N respectively.

33. What is meant by radix-2 FFT?

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

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

Decimation-in-time algorithm is used to calculate the DFT of a N-point Sequence. The idea is to break the N-point sequence into two sequences, the DFTs of which can be combined to give the DFT of the original N-point sequence. Initially the N-point sequence is divided into two N/2-point sequences xe(n) and x0(n), which have the even and odd members of x(n) respectively. The N/2 point DFTs of these two sequences are evaluated and combined to give the N point DFT. Similarly the N/2 point DFTs can be expressed as a combination of N/4 point DFTs. This process is continued till we left with 2-point DFT. This algorithm is called Decimation-in-time because the sequence x(n) is often splitted into smaller sub sequences.

35. What are the differences and similarities between DIF and DIT algorithms?
Differences:
1. For DIT, the input is bit reversal while the output is in natural order, whereas for DIF, the input is in natural order while the output is bit reversed.
2. The DIF butterfly is slightly different from the DIT butterfly, the difference being that the complex multiplication takes place after the add-subtract operation in DIF.
Similarities: Both algorithms require same number of operations to compute the DFT. Both algorithms can be done in place and both need to perform bit reversal at some place during the computation.

36. What are the applications of FFT algorithms?
1. Linear filtering
2. Correlation
3. Spectrum analysis

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

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

38. Distinguish between DFT and DTFT.

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

39. Distinguish between Fourier series and Fourier transform.

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

40. Whatis overlap-add method?

In this methodthe sizeofthe input data block xi(n) is L.Toeach data block weappendM-1zeros andperform Npointcicular convolution ofxi(n) andh(n). Since each data blockis terminatedwith M-1 zeros the lastM-1points fromeachoutput blockmust be overlappedandaddedto firstM-1points ofthe succeeding blocks.This methodis calledoverlap-addmethod.

41.Whatis overlap-save method?

In this methodthe data sequence is dividedinto N pointsections xi(n).Each section contains the last M-1 data points of the previous section followedbyLnew data points to form a data sequence oflength N=L+M-1.In circularconvolution of xi(n) with h(n) the first M-1points will not agree with the linearconvolution of xi(n) andh(n) becauseofaliasing, the remaining points will agree with linearconvolution. Hence we discardthe first (M-1) points of filteredsection xi(n) Nh(n). This process is repeated forall sections andthe filteredsections are abuttedtogether.

UNIT-III / IIR FILTER DESIGN

1.What are thedifferent types of filters based onimpulse response?

Basedon impulse response the filters are oftwotypes

1.IIR filter

2. FIR filter

- The IIRfilters are of recursive type,wherebythe presentoutput sample depends on the present input,pastinput samples andoutput samples.
- The FIR filters are ofnon recursive type,wherebythe presentoutput sample depends on thepresentinput sample andprevious input samples.

2. What are thedifferent types of filters based onfrequency response?

Basedon frequency response the filters can beclassifiedas

1. Lowpass filter
2. Highpass filter
3. Bandpass filter
4. Bandreject filter

3. Define IIR filter?

IIR filter has Infinite Impulse Response.

4. What are the various methods to design IIR filters?

- Approximation of derivatives
- Impulse invariance
- Bilinear transformation.

5. Which of the methods do you prefer for designing IIR filters? Why?

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

6. What is the main problem of bilinear transformation?

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

7. What is prewarping?

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

8. State the frequency relationship in bilinear transformation?

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

9. Where the jΩ axis of s-plane is mapped in z-plane in bilinear transformation?

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

10. Where left hand side and right hand side are mapped in z-plane in bilinear transformation?

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

11. What is the frequency response of Butterworth filter?

Butterworth filter has monotonically reducing frequency response.

12. Which filter approximation has ripples in its response?

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

13. Can IIR filter be designed without analog filters?

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

14. What is the advantage of designing IIR Filters using pole-zero plots?

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

15. Compare the digital and analog filter.

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

16. What are the advantages and disadvantages of digital filters?

Advantages of digital filters

➢High thermal stability due to absenceofresistors,inductors and capacitors.
➢Increasingthe length ofthe registerscan enhance the performance characteristics like accuracy, dynamic range, stability and tolerance.
➢The digital filters are programmable.
➢Multiplexing and adaptive filtering are possible.

Disadvantages of digital filters

➢The bandwidth of the discrete signal is limited by the sampling frequency.
➢The performance of thedigital filter depends on the hardwareused to implement the filter.

17. What is impulse invariant transformation?

The transformation ofanalog filter to digital filter without modifying the impulse response of the filter is calledimpulseinvarianttransformation.

18. Howanalogpoles are mapped to digital poles in impulseinvariant transformation?

In impulseinvariant transformation the mappingofanalog to digital polesare as follows,

➢The analogpoles on theleft half ofs-planeare mapped intothe interior ofunit circle inz-plane.
➢The analogpoles on theimaginary axis of s-plane are mapped into the unitcircle in the z-plane.
➢The analogpoles on theright halfof s-planearemapped into the exterior ofunit circle inz-plane.

19. What is the importance of polesin filter design?

The stability of a filter is related to the location ofthe 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 lieinsidethe unit circle in the z-plane.

20. Why an impulseinvariant transformationis not considered to be one-to-one?

In impulseinvariant transformation any strip of width $2\pi/T$in the s-plane for values ofs-plane in the range $(2k-1)/T \leq \Omega \leq (2k-1)\ \pi/T$is mapped into the entire z-plane. The left halfof each strip in s-plane is mapped into theinterior of unit circle in z- plane, righthalf ofeach stripin s-plane is mapped into the exteriorof unit circle in z-plane and the imaginary axis of each stripin s-plane is mapped on the unitcircle in z-plane.Hence theimpulse invariant transformation is many-to-one.

21. What is Bilinear transformation?

The bilinear transformation is conformalmapping that transforms thes-plane toz-plane. Inthis mapping the imaginary axis of s-plane is mapped intothe unitcircle inz- plane, The left half ofs-plane is mapped intointerior of unit circle in z-plane and the right halfof s-planeis mapped into exterior of unit circle inz-plane. TheBilinear mapping is a one-to-onemapping and itis accomplished when

22. Howthe order of the filter affectsthe frequency responseof Butterworthfilter.

The magnituderesponse of butterworth filter is shown in figure, from which itcan be observedthatthe magnituderesponse approaches theideal response asthe order of the filter is increased.

23. Writethe properties ofChebyshev filters.

- The magnitude responseis equiripple in the passband and monotonicin the stopband.
- The chebyshev type-1 filters are allpole designs.
- The normalized magnitude functionhas a valueof at thecutoff frequency Ω_C.
- The magnitude responseapproaches the ideal response as the value of N increases.

24. Compare the Butterworth andChebyshev Type filters.

Butterworth	Chebyshev
i. Allpole design. ii. The poles lie on acircle in s-plane. iii. The magnituderesponse is maximally flat atthe origin and monotonically decreasing function ofΩ. iv. The normalizedmagnituderesponse has a valueof1 /√2 at the cutoff frequency Ω_C. v. Only fewparametershas to be calculatedtodetermine the transfer function.	i. Allpole design. ii. The poles lie on aellipse in s-plane. iii. The magnituderesponse is equiripple in passbandand monotonically decreasing in the stopband. iv. The normalizedmagnituderesponse has a valueof1 /$\sqrt{(1+\varepsilon^2)}$atthe cutoff frequency Ω_C. v. A large number of parameters has to be calculatedtodetermine the transfer function.

UNIT-IV / FIR FILTER DESIGN

1. Whatare FIR filters?

The specificationsofthe desired filter will be given interms of idealfrequency response $H_d(w)$.The impulse response $h_d(n)$ofthe desired filter can beobtained by inverse fourier transform of$H_d(w)$,which consists ofinfinitesamples. The filters designed byselecting finite number of samples of impulseresponse arecalled FIR filters.

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

Based on impulseresponse the filters are oftwo types 1. IIR filter 2. FIR filter
The IIR filters are ofrecursive type, whereby the present outputsample depends on the presentinput, past input samples and output samples.The FIR filtersare of non recursive type, whereby the present output sample depends on the present input, andprevious outputsamples.

3. What are the different types of filter based on frequencyresponse?

The filters can be classified based on frequency response. They areI) Lowpass filter ii) High pass filteriii) Band pass filter iv)Band reject filter.

4. Distinguishbetween FIR andIIR filters.

S.No.	FIR filter	IIR filter
1.	These filterscan be easily designedto have perfectly linearphase.	These filtersdo not havelinear phase.
2.	FIR filters can be realized recursively and non-recursively.	IIR filters can be realizedrecursively.
3.	Greater flexibility to control the shape oftheirmagnituderesponse.	Less flexibility,usuallylimited to kind of filters.
4.	Errors due toroundoffnoiseare less severein FIR filters, mainly because feedback is not used.	The roundoffnoisein IIR filters are more.

5. What are the techniques of designing FIR filters?

There are three well-known methods for designingFIR filters withlinearphase. These are 1) windowsmethod 2) Frequency sampling method 3) Optimal or minimax design.

6. State the condition fora digital filter to be causal and stable.

- A digital filter is causal if its impulseresponse $h(n) = 0$for $n<0$
- A digital filter is stable if its impulseresponseis absolutely summable,

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

FIR filter is always stable becauseallits polesare at origin.

8. What are the properties ofFIR filter?

1. FIR filter is always stable.
2. A realizable filter can alwaysbe obtained.
3. FIR filter has a linearphase response.

9. Howphasedistortion and delay distortions areintroduced?

- The phase distortion is introduced when thephase characteristics ofa filter is not linear withinthe desiredfrequency band.
- The delay distortion is introduced when thedelay is notconstant within the desired frequency range.

10. Writethe steps involvedin FIR filter design.

➢Choose the desired(ideal) frequency response$H_d(w)$.
➢Take inverse fourier transformof $H_d(w)$to get $h_d(n)$.
➢Convert theinfinite duration $h_d(n)$ tofiniteduration h(n).
➢Take Z-transformof h(n) to get thetransfer function H(z) of the FIR filter.

11. What are the advantages ofFIR filters?

➢Linear phase FIR filtercan be easily designed.
➢Efficient realization of FIR filter exist asboth recursive and nonrecursive structures.
➢FIR filtersrealized nonrecursively are always stable.
➢The roundoffnoisecan be made small in nonrecursive realization ofFIR filters.

12. What are the disadvantagesof FIR filters?

➢The duration of impulseresponse should be large to realize sharp cutoff filters.
➢The non-integral delay can leadto problems in some signal processing applications.

13. What is the necessary and sufficient conditionfor the linear phase characteristicof an FIRfilter?

The necessaryandsufficient condition forthe linear phase characteristic of an FIR filter is thatthe phase function shouldbe a linear function of w, which in turn requires constant phase and groupdelay.

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

- The conditions for constant phase delayARE
- Phase delay,α = (N-1)/2(i.e., phasedelay is constant)
- Impulseresponse, h(n) =-h(N-1-n) (i.e., impulseresponseis antisymmetric)

15. Howconstant group delay &phasedelay is achieved in linear phase FIR filters?

The following conditions have tobe satisfied to achieveconstant groupdelay & phase delay.

- Phase delay,α = (N-1)/2(i.e., phasedelay is constant)
- Group delay, β = π/2 (i.e., group delay is constant)
- Impulseresponse, h(n) =-h(N-1-n) (i.e., impulseresponseis antisymmetric)

16. What are the possibletypes of impulse response for linear phase FIR filters?

There are four typesofimpulseresponse for linear phaseFIR filters
➢Symmetric impulseresponse when N is odd.
➢Symmetric impulseresponse when N is even.
➢Antisymmetricimpulseresponse when N is odd.
➢Antisymmetricimpulseresponse when N is even.

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

Thereare threewell-known design techniques of linear phase FIR filters. Theyare
➢Fourier series method and window method
➢Frequency samplingmethod.
➢Optimal filter design methods.

18. What is Gibb’s phenomenon (or Gibb’s Oscillation)?

In FIR filter design by Fourier series method the infinite durationimpulse responseis truncated to finite duration impulseresponse. Theabrupttruncation of impulseresponse introduces oscillations in the passband and stopband. This effectis known as Gibb’s phenomenon (or Gibb’s Oscillation).

19. When cascade form realization is preferred inFIR filters?

The cascade formrealization is preferred when complex zeroswith absolute magnitude less than one.

20. What are the desirable characteristics of thefrequency response ofwindow function?

The desirable characteristics of the frequency response of window function are

- The width ofthe mainlobe should be small and itshould contain as much of the total energy as possible.
- The sidelobes should decrease in energy rapidlyas w tends to π.

21. Writethe procedure for designingFIR filter using frequency-sampling method.

- Choose the desired(ideal) frequency responseH_d(w).
- Take N-samples of H_d(w)to generatethe sequence
- Take inverse DFTof to getthe impulseresponse h(n).
- The transfer function H(z) of the filter is obtainedby taking z-transformof impulseresponse.

22. What are the drawbackin FIR filter design using windowsand frequency sampling method? Howitis overcome?

The FIR filter designusing windows and frequency sampling method does nothave Precise control over thecritical frequencies suchas $w_p$and w_s.

This drawback can be overcome by designing FIR filter using Chebyshev approximation technique.In this technique an error functionis used toapproximate the ideal frequency response, in order tosatisfy the desiredspecifications.

23. Writethe characteristic features of rectangular window.

- The mainlobe width is equal to 4π/N.
- The maximum sidelobe magnitude is–13dB.
- The sidelobe magnitude does notdecrease significantly withincreasing w.

24. List the features ofFIR filter designed using rectangularwindow.

- The width ofthe transition region isrelated tothe width of the mainlobe of window spectrum.
- Gibb's oscillations are noticed in the passband and stopband.
- The attenuation in the stopband is constantand cannot be varied.

25. Why Gibb's oscillations are developed in rectangular windowand howitcan be eliminated or reduced?

The Gibb's oscillations in rectangularwindow are due to the sharptransitions from 1 to 0 attheedges of window sequence.

These oscillations can be eliminatedor reducedby replacingthe sharptransition by gradualtransition. This isthe motivation for development of triangularand cosine windows.

26. List the characteristicsof FIR filters designedusing windows.

➢The width ofthe transition band depends on thetype of window.

➢The width ofthe transition band canbe made narrow by increasing the value of N where N is the lengthof the window sequence.

➢The attenuation in the stop band is fixed for a given window, except in case of Kaiser window where itis variable.

27. Compare the rectangular windowand hanning window.

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

28. Compare the rectangular windowand hammingwindow.

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

29. Writethe characteristic features of hanning windowspectrum.

➢The mainlobe width is equal to 8π/N.

➢The maximum sidelobe magnitude is–41dB.

➢The sidelobe magnitude remains constant forincreasing w.

30. What is the mathematical probleminvolved inthe designof windowfunction?

Themathematical probleminvolved inthe design of window function(or sequence) isthat of finding a time-limited function whose Fourier Transformbest approximates a band limited function. The approximation should be suchthat the maximum energy is confined to mainlobe for a given peaksidelobe amplitude.

31. List the desirable features of KaiserWindowspectrum.

➢The width ofthe mainlobe and the peak sidelobeare variable.
➢The parameter α inthe KaiserWindow function is an independent variable that can be varied to control the sidelobelevels with respectto mainlobe peak.
➢The width ofthe mainlobe in the window spectrumcan be varied by varying the length N of the window sequence.

32. Compare the hamming window and Kaiserwindow.

HammingWindow	Kaiser Window
i)The width of mainlobe in window spectrumis 8π/N ii) The maximumsidelobe magnitude in window spectrumis–41dB. iii) In window spectrum the sidelobe magnitude remains constant. iv)In FIR filter designed using hamming window the minimum stopband attenuationis 44dB.	i)The width of mainlobe in window spectrumdepends on the values of α& N. ii) The maximumsidelobe magnitude with respect to peak of mainlobe isvariable using the parameter α. iii) In window spectrum the sidelobe magnitude decreases with increasing w. iv)In FIR filter designed using Kaiser window the minimum stopband attenuationis variable and depends on the value of α.

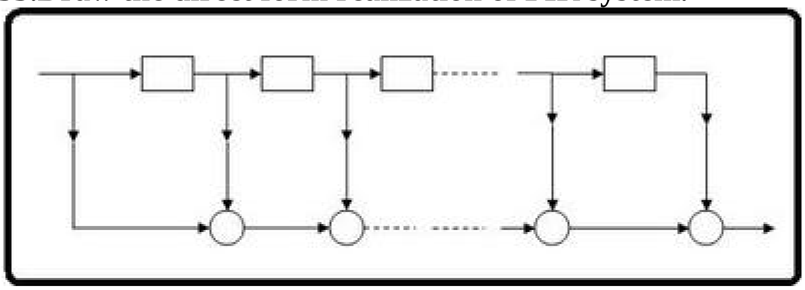
33.Whatis the principle ofdesigning FIRfilter using frequency sampling method?

In frequencysamplingmethodthe desiredmagnitude response is sampled anda linearphase response is specified.The samples of desired frequency response are identifiedas DFTcoefficients.The filtercoefficients are then determinedas the IDFT ofthis set of samples.

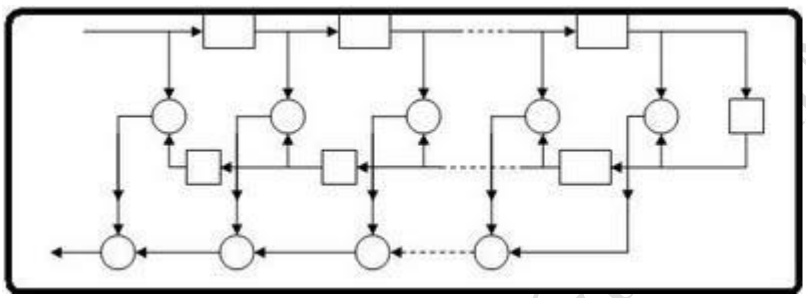
34. For what type of filters frequency sampling methodis suitable?

Frequencysampling methodis attractive fornarrowbandfrequencyselective filters where onlya fewofthe samples of the frequency response are non zero.

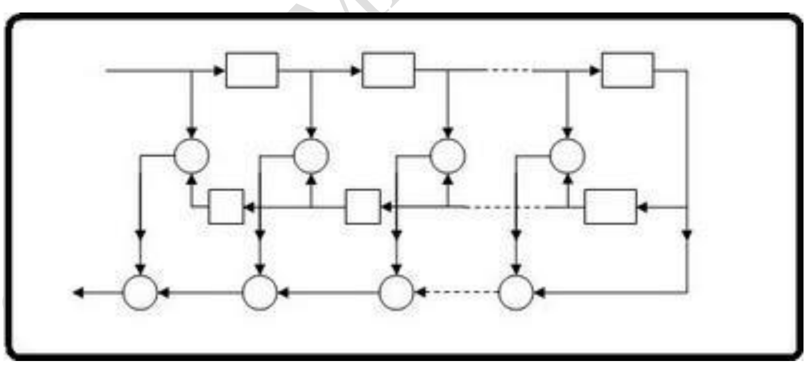
35.Draw the direct form realization of FIR system.



36.Draw the direct form realization of a linearPhase FIR system forNeven.



37.Draw the direct form realization of a linearPhase FIR system forNodd



UNIT-V / FINITE WORD LENGTH

1. whatare the different types ofarithmetic indigital systems.?

There are threetypes ofarithmetic used indigital systems. Theyare fixedpointarithmetic, floating point ,block floating point arithmetic.

2.Whatis meantby fixedpointnumber?

In fixedpointnumberthe position ofa binarypointis fixed.The bittothe right representthe fractional part andthose to the left is integer part.

3.What are thedifferent types of fixed point arithmetic?

Dependingon the negative numbers are representedthere are threeforms offixedpointarithmetic.They are sign magnitude,1's complement,2's complement

4. Whatis meantby signmagnitude representation?

Forsign magnitude representation the leadingbinary digit is usedto representthe sign. Ifitis equal to 1the number is negative,otherwiseitis positive.

5. Whatis meantby 1's complement form?

In 1,s complementformthe positive number isrepresentedas in the sign magnitude form. To obtain the negative ofthe positive number ,complement all the bits ofthe positive number.

6. Whatis meantby 2's complement form?

In 2's complementformthe positive number isrepresentedas in the sign magnitude form. To obtain the negative ofthe positive number ,complement all the bits ofthe positive number and add1to the LSB.

7. Whatis meantby floating pintrepresentation?

In floatingpointform the positive number is representedas F=2CM,where is mantissa,is a fraction such that1/2<M<1andCthe exponent can be eitherpositive ornegative.

8. What are the advantages offloating pintrepresentation?

1.Large dynamic range2.overflowis unlikely.

9.Whatare the quantization errors due tofinitewordlengthregisters indigital filters?

1.Input quantization errors
2.Coefficient quantization errors
3.Productquantization errors

10.Whatis inputquantization error?.

The filtercoefficients are computedto infiniteprecision intheory.But in digital computationthe filter coefficients are representedinbinaryandare storedinregisters.Ifa bbitregisteris usedthefilter coefficients must beroundedortruncatedto bbits ,whichproduces an error.

11.Whatis productquantization error?.

The product quantization errors arise at the out put ofthe multiplier.Multiplication of a bbitdata with a b bitcoefficient results a product having2b bits.Since a bbitregisteris usedthe multiplieroutput will be roundedortruncatedto bbits whichproduces theerror.

12. Whatis inputquantization error?.

The input quantizationerrors arise due to A/Dconversion.

13.What are thedifferentquantization methods?

1. Truncation and2.Rounding

14.Whatis truncation?

Truncation isa process of discardingall bits lesssignificant than LSB that is retained

15. Whatis Rounding?

Roundinga number tob bits is accomplished bychoosing a roundedresult as the bbitnumber closestnumberbeing unrounded.

16.What are thetwotypes oflimitcycle behavior ofDSP?.

1. Zero limit cycle behavior 2.Over flowlimit cycle behavior

17.What are the methods to preventoverflow?

1. Saturationarithmeticand 2.Scaling

18.State some applications of DSP?

Speechprocesses,Image processing,Radar signal processing.