TWO MARK QUESTIONS WITH ANSWERS

UNIT-I

1. What is DFT?

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

2. Define N point DFT.

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

3. What is DFT of unit impulse δ(n)?

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

4. List the properties of DFT.

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

5. 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.

6. 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

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7. 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.

8. 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)

9. 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_{Nkl}

10. **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.

11. 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.

12. What is the computational complexity using FFT algorithm?
1. Complex multiplications = $N/2 \log_2 N$
2. Complex additions = $N \log_2 N$

13. 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.

14. What is zero padding? What are its uses?
Let the sequence x (n) has a length L. If we want to find the N point DFT (N>L) of the sequence x (n). This is known as zero padding. The uses of padding a sequence with zeros are
(i) We can get 'better display' of the frequency spectrum.
(ii) With zero padding, the DFT can be used in linear filtering.

15. Why FFT is needed?
The direct evaluation of the DFT using the formula requires N_2 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 $N/2\log_2 N$. If N=16, the number of complex multiplications required for direct evaluation of DFT is 256, whereas using DFT only 32 multiplications are required.

16. 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 N_2=642=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

17. What is the main advantage of FFT?
FFT reduces the computation time required to compute discrete Fourier transform.

18. Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with using FFT algorithm with 32-point sequence.
For N-point DFT the number of complex multiplications needed using FFT algorithm is $N/2 \log_2 N$. For N=32, the number of the complex multiplications is equal to $32/2\log_2 32$=16*5=80.

19. What is FFT?

The fast Fourier transforms (FFT) is an algorithm used to compute the DFT. It makes use of the Symmetry and periodically properties of twiddles factor W_{KN} 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.

20. 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 $N \log_2 N$ and $N/2 \log_2 N$ respectively.

21. 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-s FFT algorithm

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

Decimation-in-time algorithm is used to calculate the DFT of a N-point Sequence. The idea is to break the N-point sequence into two sequences, the DFTs of which can be combined to give the DFT of the original N-point sequence. Initially the N-point sequence is divided into two N/2-point sequences $x_e(n)$ and $x_0(n)$, 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.

23. 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. Bot algorithms can be done in place and both need to perform bit reversal at some place during the computation.

24. What are the applications of FFT algorithms?

1. Linear filtering
2. Correlation
3. Spectrum analysis

25. 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.

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

27. Distinguish between Fourier series and Fourier transform.
S.No.
Gives the harmonic content of a periodic time function. Discrete frequency spectrum
Gives the frequency information for an aperiodic signal. Continuous frequency spectrum

UNIT-II

1) Define IIR filter?
IIR filter has Infinite Impulse Response.

2) What are the various methods to design IIR filters?

1. Approximation of derivatives
2. Impulse invariance
3. Bilinear transformation.

3) Which of the methods do you prefer for designing IIR filters? Why?
Bilinear transformation is best method to design IIR filter, since there is no aliasing in it.

4) What is the main problem of bilinear transformation?
Frequency warping or nonlinear relationship is the main problem of bilinear transformation.

5) What is prewarping?
Prewarping is the method of introducing nonlinearely in frequency relationship to compensate warping effect.

6) State the frequency relationship in bilinear transformation?

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

7) Where the jΩaxis of s-plane is mapped in z-plane in bilinear transformation?
The jΩaxis of s-plane is mapped on the unit circle in z-plane in bilinear transformation

8) Where left hand side and right hand side are mapped in z-plane in bilinear transformation?
Left hand side -- Inside unit circle Right hand side – Outside unit circle

9) What is the frequency response of Butterworth filter?
Butterworth filter has monotonically reducing frequency response.

10) Which filter approximation has ripples in its response?
Chebyshev approximation has ripples in its pass band or stop band.

11) Can IIR filter be designed without analog filters?
Yes. IIR filter can be designed using pole-zero plot without analog filters

12) What is the advantage of designing IIR Filters using pole-zero plots?
The frequency response can be located exactly with the help of poles and zeros.

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

14) What are the advantages and disadvantages of digital filters?
Advantages of digital filters

1. High thermal stability due to absence of resistors, inductors and capacitors.
2. Increasing the length of the registers can enhance the performance characteristics like accuracy, dynamic range, stability and tolerance.
3. The digital filters are programmable.
4. Multiplexing and adaptive filtering are possible.

Disadvantages of digital filters

1. The bandwidth of the discrete signal is limited by the sampling frequency.
2. The performance of the digital filter depends on the hardware used to implement the filter.

15) What is impulse invariant transformation?
The transformation of analog filter to digital filter without modifying the impulse response of the filter is called impulse invariant transformation.

16) Obtain the impulse response of digital filter to correspond to an analog filter with impulse response $h_a(t) = 0.5\ e^{-2t}$ and with a sampling rate of 1.0kHz using impulse invariant method.

17) How analog poles are mapped to digital poles in impulse invariant transformation?
In impulse invariant transformation the mapping of analog to digital poles are as follows,
1. The analog poles on the left half of s-plane are mapped into the interior of unit circle in zplane.
2. The analog poles on the imaginary axis of s-plane are mapped into the unit circle in the zplane.

3.The analog poles on the right half of s-plane are mapped into the exterior of unit circle in z plane.

18) 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.

19) Why an impulse invariant transformation is not considered to be one-to-one?
In impulse invariant transformation any strip of width $2\pi/T$ in the s-plane for values of s-plane in
the range (2k-1)/T □Ω □(2k-1) π/T is mapped into the entire z-plane. The left half of each strip in s-plane is mapped into the interior of unit circle in z-plane, right half of each strip in s-plane is mapped into the exterior of unit circle in z-plane and the imaginary axis of each strip in s-plane is mapped on the unit circle in z-plane. Hence the impulse invariant transformation is many-to-one.

20) What is Bilinear transformation?
The bilinear transformation is conformal mapping that transforms the s-plane to z-plane. In this mapping the imaginary axis of s-plane is mapped into the unit circle in z-plane, The left half of s plane is mapped into interior of unit circle in z-plane and the right half of s-plane is mapped into exterior of unit circle in z-plane. The Bilinear mapping is a one-to-one mapping.

21) How the order of the filter affects the frequency response of Butterworth filter.
The magnitude response of butterworth filter is shown in figure, from which it can be observed that the magnitude response approaches the ideal response as the order of the filter is increased.

22) Write the properties of Chebyshev type –1 filters.
1.The magnitude response is equiripple in the passband and monotonic in the stopband.

2.The chebyshev type-1 filters are all pole designs.

3.The normalized magnitude function has a value of at the cutoff frequency Ω_c.

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

23) Compare the Butterworth and Chebyshev Type-1 filters.
Butterworth

i. All pole design.
ii. The poles lie on a circle in s-plane.
iii. The magnitude response is maximally flat at the origin and monotonically decreasing function of Ω
iv. The normalized magnitude response has a value of 1 / □2 at the cutoff frequency Ω
v. Only few parameters has to be calculated to determine the transfer function.

Chebyshev Type - 1

i. All pole design.

ii. The poles lie on a ellipse in s-plane.

iii. The magnitude response is equiripple in passband and monotonically decreasing in the stopband.

iv. The normalized magnitude response has a value of $1 / \square(1+\square_{2})$ at the cutoff frequency Ω_c.

v. A large number of parameters has to be calculated to determine the transfer function.