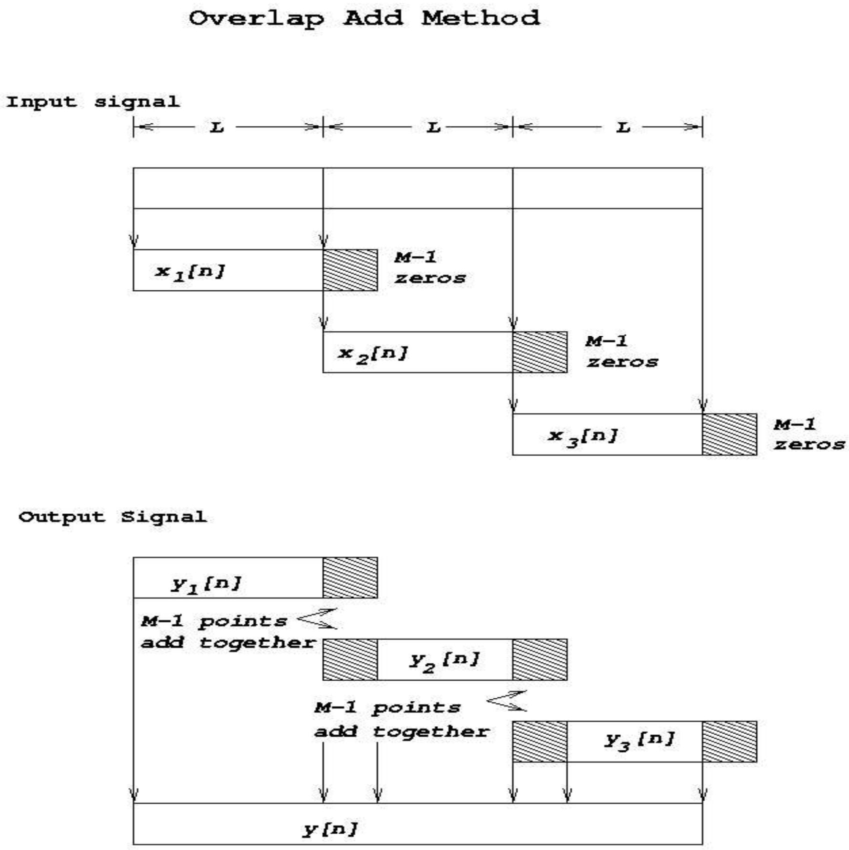


Two Mark Questions and Answers:

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. This is affected by means of transducer. (e.g.) microphone, photocell.

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: The 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 modelled as completely specified functions of time.
Random signal: Random signal is a signal about which there is uncertainty before its actual occurrence. Such signal may be viewed as group of signals with each signal in the ensemble having different wave forms .(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 powered 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, Unit step function, Unit impulse function, Ramp function, Exponential function and Sinusoidal function

8. What are the properties of a system?
Stability
Memory
Invertibility
Time invariance

Linearity

9. What is memory system and memory less system?

A system is said to be memory system if its output signal at any time depends on the past values of the input signal. Circuit with inductors capacitors are examples of memory system..

A system is said to be memory less system if the output at any time depends on the present values of the input signal. An electronic circuit with resistors is an example for memory less system.

10. What is an invertible system?

A system is said to be invertible system if the input of the system can be recovered from the system output. The set of operations needed to recover the input as the second system connected in cascade with the given system such that the output signal of the second system is equal to the input signal applied to the system.

11. 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 idenditical shift in the output signal. This implies that a time invariant system responds idenditically no matter when the input signal is applied.

12. What is SISO system and MIMO system?

A control system with single input and single output is referred to as single input single output system. When the number of plant inputs or the number of plant outputs is more than one the system is referred to as multiple input output system. In both the case, the controller may be in the form of a digital computer or microprocessor in which we can speak of the digital control systems.

13. 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 $x(n+N)\neq x(n)$

14. Determine the convolution sum of two sequences x (n) = {3, 2, 1, 2} and h (n) = {1, 2, 1, 2}

y(n) = {3,8,8,12,9,4,4}

15.State the classification of systems.

- Static and dynamic system.
- Time invariant and time variant system.
- Causal and anticausal system.
- Linear and Non-linear system.
- Stable and Unstable system.

16.Define causal and anticausal 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.

17.Define stable and unable system.
A system is said to be stable if we get bounded output for bounded input.

18.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 duration L+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.

19.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.

20.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.

21.What are the two methods used for the sectional convolution?
The two methods used for the sectional convolution are
1) overlap-add method and 2)overlap-save method.

22.What is overlap-add method?
In this method the size of the input data block xi(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.

23.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 xi(n) N h(n). This process is repeated for all sections and the filtered sections are abutted together.

24. Differentiate DTFT and DFT
DTFT output is continuous in time where as DFT output is Discrete in time.

25. Differentiate between DIT and DIF algorithm
<u>DIT</u> – Time is decimated and input is bit reversed format output in natural order
<u>DIF</u> – Frequency is decimated and input is natural order output is bit reversed format.

26.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.
27.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 DFT computation time. It is based on the fundamental principle of decomposing the computation of DFT of a sequence of length N into successively smaller DFTs.

28.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 radix-2 FFT are N log2 N and N/2 log2 N respectively,.

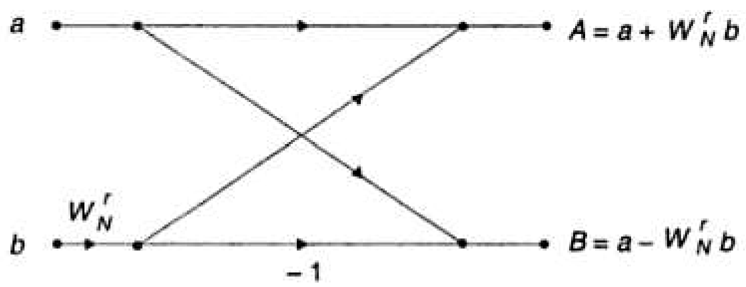
29.What is meant by radix-2 FFT?
The FFT algorithm is most efficient in calculating N point DFT. If the number of

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-2 algorithm.

30.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.

31.What is 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.

31.Draw the basic butterfly diagram of DIT algorithm.
The basic butterfly diagram for DIT algorithm is Where a and b are inputs and A and B are the outputs.



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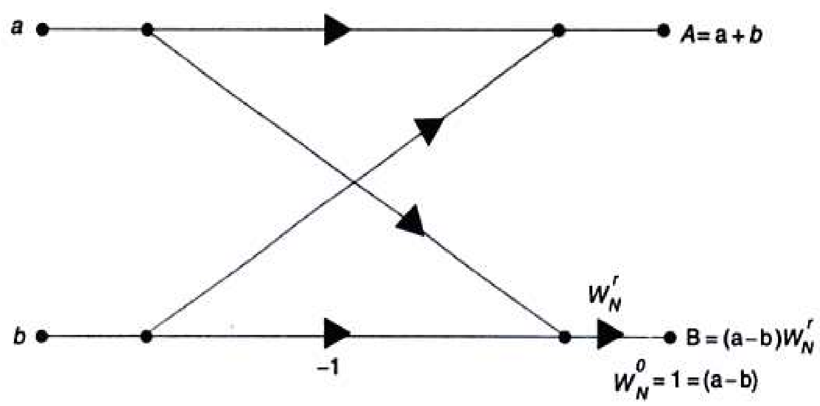

33.What are the applications of FFT algorithm?
The applications of FFT algorithm includes
1) Linear filtering
2) Correlation
3) Spectrum analysis

34.Why the computations in FFT algorithm is said to be in place?
Once the butterfly operation is performed on a pair of complex numbers (a,b) to produce (A,B), there is no need to save the input pair. We can store the result (A,B) in the same locations as (a,b). Since the same storage locations are used throughout the computation we say that the computations are done in place.

35.Distinguish between linear convolution and circular convolution of two sequences.
Linear convolution:
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. It can be used to find the response of a linear filter.
Zero padding is not necessary to find the response of a linear filter.
Circular convolution:
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. It cannot be used to find the response of a filter. Zero padding is necessary to find the response of a filter.

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

Overlap-save method
1.In this method the size of the input data block is N=L+M-1
2.Each data block consists of the last M-1 data points of the previous data block followed by L new data points
3.In each output block M-1 points are corrupted due to aliasing as circular convolution is employed
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
Overlap-add method
In this method the size of the input data block is L
Each data block is L points and we append M-1 zeros to compute N point DFT
In this no corruption due to aliasing as linear convolution is performed using circular convolution
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

37.What are the differences and similarities between DIF and DIT algorithms?

Differences:

1)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.

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.

Summary:

DFT is an important mathematical tool which can be used for the software implementation of certain digital signal processing algorithms .DFT gives a method to transform a given sequence to frequency domain and to represent the spectrum of the sequence using only k frequency values, where k is an integer that takes N values, k=0, 1, .,..N-1.

The (DFT discrete Fourier transform) is one of the specific forms of Fourier analysis. As such, it transforms one function into another, which is called the frequency domain representation, or simply the DFT, of the original function (which is often a function in the time domain). But the DFT requires an input function that is discrete and whose non-zero values have a limited (finite) duration. Such inputs are often created by sampling a continuous function, like a person's voice.