n	h(n)
0	0.0397
1	-0.048
2	-0.0345
3	0.0659
4	0.0315
5	-0.1074
6	-0.0299
7	0.31876
8	0.5294
9	0.31876
10	-0.0299
11	-0.1074
12	0.0315
13	0.0659
14	-0.0345
15	-0.0488
16	0.0397

Two Marks Questions and Answers:

1) 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.

2.What is filter?

Filter is a frequency selective device , which amplify particular range of frequencies and attenuate particular range of frequencies.

3. 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.

4. What is FIR filters?

The specifications of the desired filter will be given in terms of ideal frequency response Hd(w). The impulse response hd(n) of the desired filter can be obtained by inverse fourier transform of Hd(w), which consists of infinite samples. The filters designed by selecting finite number of samples of impulse response are called FIR filters.

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

Based on frequency response the filters can be classified as

1. Low pass filter
2. High pass filter
3. Band pass filter
4. Band reject filter

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

Symmetric condition $h(n)=h(N-1-n)$.The ant symmetrical impulse response can be used to design Hilbert transforms and differentiators.

7. For what kind of application, the symmetrical impulse response can be used?

Anti symmetric condition $h(n)=-h(N-1-n)$.The impulse response, which is symmetric having odd number of samples can be used to design all types of filters, i.e, low pass, high pass, band pass and band reject. The symmetric impulse response having even number of samples can be used to design lo pass and band pass filter.

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

Advantages:

1. FIR filters have exact linear phase.
2. FIR filters are always stable.
3. FIR filters can be realized in both recursive and non recursive structure.
4. Filters with any arbitrary magnitude response can be tackled using FIR sequence.

Disadvantages:

1. 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.
2. Large storage requirement is requirement
3. Powerful computational facilities required for the Implementation.

9. Distinguish between FIR and IIR filters.

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

10. What are the design techniques of designing FIR filters?

There are three well known methods for designing FIR filters with linear phase. They are

(1) Window method

(2) Frequency sampling method

(3) Optimal or minimax design.

11. State the condition for a digital filter to be causal and stable.

A digital filter is causal if its impulse response h(n) = 0 for n<0.A digital filter is stable if its impulse response is absolutely summable,

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

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

13. What are the properties of FIR filter?

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

14. How phase distortion and delay distortions are introduced?

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

15. Write the steps involved in FIR filter design.

- Choose the desired (ideal) frequency response Hd(w).
- Take inverse fourier transform of Hd(w) to get hd(n).
- Convert the infinite duration hd(n) to finite duration h(n).
- Take Z-transform of h(n) to get the transfer function H(z) of the FIR filter.

16. What is the necessary and sufficient condition for the linear phase characteristic of an FIR

filter?

The necessary and sufficient condition for the linear phase characteristic of an FIR filter is that the phase function should be a linear
function of w, which in turn requires constant phase and group delay.

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

The conditions for constant phase delay ARE Phase delay, $\alpha = (N-1)/2$ (i.e., phase delay is constant) Impulse response, $h(n) = -h(N-1-n)$ (i.e., impulse response is anti symmetric)

18. How constant group delay & phase delay is achieved in linear phase FIR filters?

The following conditions have to be satisfied to achieve constant group delay & phase delay.
Phase delay, $\alpha = (N-1)/2$ (i.e., phase delay is constant)
Group delay, $\beta = \pi/2$ (i.e., group delay is constant)
Impulse response, $h(n) = -h(N-1-n)$ (i.e., impulse response is antisymmetric)

19. What are the possible types of impulse response for linear phase FIR filters?

There are four types of impulse response for linear phase FIR filters

- Symmetric impulse response when N is odd.
- Symmetric impulse response when N is even.
- Antisymmetric impulse response when N is odd.
- Antisymmetric impulse response when N is even.

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

There are three well-known design techniques of linear phase FIR filters. They are

- Fourier series method and window method
- Frequency sampling method.
- Optimal filter design methods.

21. What is the necessary and sufficient condition for linear phase characteristic in FIR filter?

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

22. When cascade form realization is preferred in FIR filters?

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

23. What are the desirable characteristics of the window function?

The desirable characteristics of the window are
1. The central lobe of the frequency response of the window should contain most of the energy and should be narrow.
2. The highest side lobe level of the frequency response should be small.

3 .The side lobes of the frequency response should decrease in energy
rapidly as ω tends to π .

24. 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 identified as DFT coefficients. The filter coefficients are then determined as the IDFT of this set of samples.

25. 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.

26. Write the procedure for designing FIR filter using frequency-sampling method.

- Choose the desired (ideal) frequency response Hd(w).
- Take N-samples of Hd(w) to generate the sequence
- Take inverse DFT of to get the impulse response h(n).
- The transfer function H(z) of the filter is obtained by taking z-transform of impulse
- response.

27. What are the drawback in FIR filter design using windows and frequency sampling method? How it is overcome?

The FIR filter design using windows and frequency sampling method does not have Precise control over the critical frequencies such as wp and ws. This drawback can be overcome by designing FIR filter using Chebyshev approximation technique. In this technique an error function is used to approximate the ideal frequency response, in order to satisfy the desired specifications.

28. Write the characteristic features of rectangular window.

- The main lobe width is equal to $4\pi/N$.
- The maximum sidelobe magnitude is –13dB.
- The sidelobe magnitude does not decrease significantly with increasing w.

29. List the features of FIR filter designed using rectangular window.

- The width of the transition region is related to the width of the mainlobe of window spectrum.
- Gibb's oscillations are noticed in the passband and stopband.
- The attenuation in the stopband is constant and cannot be varied.

30. Why Gibb's oscillations are developed in rectangular window and how it can be eliminated or
reduced?

The Gibb's oscillations in rectangular window are due to the sharp transitions from 1 to 0 at the edges of window sequence.These oscillations can be eliminated or reduced by replacing the sharp transition by gradual transition. This is the motivation for development of triangular and cosine windows.

31. List the characteristics of FIR filters designed using windows.
   - The width of the transition band depends on the type of window.
   - The width of the transition band can be made narrow by increasing the value of N where N is the length of the window sequence.
   - The attenuation in the stop band is fixed for a given window, except in case of Kaiser window where it is variable.

32. Compare the rectangular window and hamming window.

Rectangular window	Hamming Window
i) The width of main lobe in window spectrum is $4\pi/N$ ii) The maximum side lobe magnitude in window spectrum is –13dB. iii) In window spectrum the side lobe magnitude slightly decreases with increasing w. iv) In FIR filter designed using rectangular window the minimum stop band attenuation is 22dB.	i) The width of main lobe in window spectrum is $8\pi/N$ ii) The maximum side lobe magnitude in window spectrum is –31dB. iii) In window spectrum the side lobe magnitude decreases with increasing w. iv) In FIR filter designed using hamming window the minimum stop band attenuation is 44dB.

33. Write the characteristic features of hanning window spectrum.
   - The mainlobe width is equal to $8\pi/N$.
   - The maximum sidelobe magnitude is –41dB.
   - The sidelobe magnitude remains constant for increasing w.

34. What is the mathematical problem involved in the design of window function?

The mathematical problem involved in the design of window function(or sequence) is that of finding a time-limited function whose Fourier Transform best approximates a band limited function. The approximation should be such that the maximum energy is confined to mainlobe for a given peak sidelobe amplitude.

35.What are the disadvantage of Fourier series method ?

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

36. What is Gibbs phenomena?
OR
What are Gibbs oscillations?

One possible way of finding an FIR filter that approximates H(ejω)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.

Summary:

A digital filter is a mathematical algorithm implemented in hardware/software that operates on a digital input to produce a digital output. Digital filters often operate on digitized analog signals stored in a computer memory. Digital filters play very important roles in DSP. Compared with analog filters, they are preferred in a number of applications like data compression, speech processing, image processing, etc.,

FIR filters are employed in filtering problems where linear phase characteristics within the pass band of the filter are required. If this is not required, either an IIR or an FIR filter may be employed. An IIR filter has lesser number of side lobes in the stop band than an FIR filter with the same number of parameters. For this reason if some phase distortion is tolerable, an IIR filter is preferable. Also, the implementation of an IIR filter involves fewer parameters, less memory requirements and lower computational complexity.