1. Define LTI causal system? (NOV 2007)
   A LTI system is causal if and only if $h(n) = 0$ for $n<0$. This is the sufficient and necessary condition for causality of the system.

2. Define LTI stable system?
   The bounded input $x(n)$ produces bounded output $y(n)$ in the LTI system only if,
   \[ \sum_{n=-\infty}^{\infty} |h(n)| < \infty \]. When this condition is satisfied, the system will be stable.

3. Define system function?
   $H(z) = \frac{Y(z)}{X(Z)}$ is called system function. It is the z transform of the unit sample response $h(n)$ of the system.

4. How unit sample response of discrete time system is defined?
   The unit step response of the discrete time system is output of the system to unit sample sequence. i.e., $T[\delta(n)] = h(n)$. Also $h(n) = z^{-1}\{H(z)\}$.

5. A causal DT system is BIBO stable only if its transfer function has ________.
   Ans: A causal DT system is stable if poles of its transfer function lie within the unit circle.

6. If $u(n)$ is the impulse response response of the system, What is its step response?
   Here $h(n) = u(n)$ and the input is $x(n) = u(n)$.
   Hence the output $y(n) = h(n) \ast x(n) = u(n) \ast u(n)$

7. Convolve the two sequences $x(n) = \{1,2,3\}$ and $h(n) = \{5,4,6,2\}$.
   Ans: $y(n) = \{5,14,29,26,22,6\}$

8. Determine the range of values of the parameter ‘a’ for which the linear time Invariant system with impulse response $h(n) = a^n u(n)$ is stable?
   Ans: $H(z) = \frac{1}{1-az^{-1}}$, There is one pole at $z=a$. The system is stable, if all its poles. i.e., within the unit circle. Hence $|a| < 1$ for stability.
9. What are the basic elements of Block diagram?
   1. Adder
   2. Multiplier
   3. Unit delay

10. Write the state model for DT system?
    \[ X_{K+1} = F + G_{UK} \]
    \[ Y_K = H_X + J_{UK} \]

11. Define Transfer function in DT System.
    It is defined as the ratio of Z transform of the system output to the Z-transform of system input.

    The response to unit impulse input is called the impulse response. The inverse Z-transform of \( H(Z) \) gives \( h(n) \).

    A signal is a function of one or more independent variables which contain some information. Eg: Radio signal, TV signal, Telephone signal etc.

    A system is a set of elements or functional block that are connected together and produces an output in response to an input signal.
    Eg: An audio amplifier, attenuator, TV set etc.

15. Define CT signals.
    Continuous time signals are defined for all values of time. It is also called as an analog signal and is represented by \( x(t) \).
    Eg: AC waveform, ECG etc.

    Discrete time signals are defined at discrete instances of time. It is represented by
x(n). Eg: Amount deposited in a bank per month.

17. Define unit step, ramp and delta functions for CT.
Unit step function is defined as
\[ U(t) = \begin{cases} 1 & \text{for } t \geq 0 \\ 0 & \text{otherwise} \end{cases} \]
Unit ramp function is defined as
\[ r(t) = \begin{cases} t & \text{for } t \geq 0 \\ 0 & \text{for } t < 0 \end{cases} \]
Unit delta function is defined as
\[ \delta(t) = \begin{cases} 1 & \text{for } t = 0 \\ 0 & \text{otherwise} \end{cases} \]

18. Define random signals.
A random signal is one which cannot be represented by any mathematical equation. Eg: Noise generated in electronic components, transmission channels, cables etc.

19. State the classification or characteristics of CT and DT systems.
The DT and CT systems are according to their characteristics as follows
(i). Linear and Non-Linear systems
(ii). Time invariant and Time varying systems.
(iii). Causal and Non-causal systems.
(iv). Stable and unstable systems.
(v). Static and dynamic systems.
(vi). Inverse systems.

20. Define linear and non-linear systems.
A system is said to be linear if superposition theorem applies to that system. If it does not satisfy the superposition theorem, then it is said to be a nonlinear system.

A system is said to be a causal if its output at anytime depends upon present and past
inputs only. A system is said to be non-causal system if its output depends upon future inputs also.

22. Define time invariant and time varying systems.
A system is time invariant if the time shift in the input signal results in corresponding time shift in the output. A system which does not satisfy the above condition is time variant system.

23. Define stable and unstable systems. (MAY 2006)
When the system produces bounded output for bounded input, then the system is called bounded input, bounded output stable. A system which doesnot satisfy the above condition is called a unstable system.

24. Define Static and Dynamic system.
A system is said to be static or memoryless if its output depends upon the present input only. The system is said to be dynamic with memory if its output depends upon the present and past input values.

It is the process of converting CT signals to DT signals. Sampling frequency must be atleast twice the highest frequency.

If the sampling rate is not enough, then low frequency signals will pose as high frequency components which is called as aliasing error.

27. Define Nyquist rate.
The minimum sampling rate at which the signals are sampled. Fs = 2fm.

28. What are the used to avoid aliasing? (MAY 2006) (APR 2008)
1) Ideal low pas filter
2) Bandlimiting the signal
29. Define Sampling theorem.
A band limited signal of finite energy, which has no frequency components higher than \( w \) Hz, is completely specifying the values of signal at instants of time separated by \( 1/2W \) sec.

30. What is Nyquist interval?
The time interval between any two adjacent samples when sampling rate is Nyquist rate.

12 marks:
1. Discuss the classification of DT and CT systems with examples.
2. Problems on the properties & classifications of signals & systems
Find whether the following signals are periodic or not (APR 2008, MAY 2006).
   a. \( x(t) = 2\cos(10t+1) - \sin(4t-1) \)
   b. \( x(t) = 3\cos(4t) + 2\sin(\pi t) \)

Check whether the following system is
1. Static or dynamic
2. Linear or non-linear
3. Causal or non-causal
4. Time invariant or variant
   \( y(n) = \text{sgn}[x(n)] \)

3. Explain sampling theorem.
4. Explain the various types of sequence and sequence representation.
5. Realization of LTI CT system using direct form I and II structures (MAY 2006).
Unit : 2. DIGITAL FILTER STRUCTURES

1. 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 $H(\omega)$ are called FIR filters.

2. Write the steps involved in FIR filter design.
   - Choose the desired frequency response $H_d(\omega)$
   - Take the inverse Fourier transform and obtain $h_d(n)$
   - Convert the infinite duration sequence $h_d(n)$ to $h(n)$

3. What are advantages of FIR filter? (Oct 98)
   - 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.

4. What are 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.

5. What is the necessary and sufficient condition for the linear phase characteristic of a FIR filter?
The phase function should be a linear function of $\omega$, which in turn requires constant group delay and phase delay.

6. List the well known design technique for linear phase FIR filter design? (Oct98,Apr98)
   - Fourier series method and window method
   - Frequency sampling method.
   - Optimal filter design method.
7. 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. low pass, high pass, band pass and band reject.
   - The symmetric impulse response having even number of samples can be used to design low pass and band pass filter.

8. What is the reason that FIR filter is always stable?
   FIR filter is always stable because all its poles are at the origin.

9. What condition on the FIR sequence \( h(n) \) are to be imposed in order that this filter can be called a linear phase filter? NOV/DEC 2005
   The conditions are (i) Symmetric condition \( h(n) = h(N-n) \)
   (ii) Antisymmetric condition \( h(n) = -h(N-n) \)

10. 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 antisymmetrical, satisfying the condition \( h(n) = -h(N-n) \). The frequency response of FIR filter will have constant group delay and not the phase delay.

11. What are the properties of FIR filter? (Apr 98)
    - FIR filter is always stable.
    - A realizable filter can always be obtained.

12. When cascade from realization is preferred in FIR filters?
    The cascade from realization is preferred when complex zeros with absolute magnitude less than one.

    NOV/DEC 2009
    One possible way of finding an FIR filter that approximates \( H(e^{j\omega}) \) would be to truncate the infinite Fourier series at \( n = \omega(N-1/2) \). 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. What is the necessary and sufficient condition for linear phase characteristics in FIR filter? MAY/JUNE 2006
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., \( h(n) = h(N-n) \) where \( N \) is the duration of the sequence.

19. 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.

20. For what type of filters frequency sampling method is suitable? NOV/DEC 2005
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.

21. What are the requirements of analog filters?
- Transfer function should be rational function
- Coefficient should be real
- Poles in the left half

22. Mention the disadvantages of digital filters.
- Bandwidth is limited
- Performance depend on hardware

25. Describe the features of IIR filters.
- Physically realizable
- Desired characteristics for magnitude

26. Define FIR.
This type of system has an impulse response which is zero outside a finite time interval

27. Define IIR
This type of system has an impulse response of infinite time interval.

28. Mention the features of IIR filters
   - No linear phase
   - Have desired characteristics for magnitude response only

29. Mention the classification of filter based on frequency response.
   - Low pass
   - High pass
   - Band pass
   - Band stop

30. Mention the advantages of digital filters
   - High thermal stability
   - Accurate
   - Easily programmable
   - Filtering is possible

12 MARKS:
1. Prove that an FIR filter has linear phase if the unit sample response satisfies the
   Condition \( h(n) = \pm h(M-n), n = 0,1,\ldots, M-1 \). Also discuss symmetric and anti symmetric
cases of FIR filter. MAY/JUNE 2006
2. Explain the need for the use of window sequence in the design of FIR filter. Describe
   the window sequence generally used and compare the properties.
3. Explain the type 1 design of FIR filter using Frequency sampling technique.
4. A LPF has the desired response given below APRIL/MAY 2007
   \[ H(\exp(j\omega)) = \exp(-3j\omega) \text{ for } 0 \leq \omega \leq \pi/2 \]
   \[ 0 \text{ for } \pi/2 \leq \omega \leq \pi \]
   Determine the filter coefficients \( h(n) \) for \( M=7 \) using frequency sampling technique.
5. Explain the principle and procedure for designing FIR filter using rectangular window.
6. Design a filter with \( H_d(\exp(-j\omega)) = \exp(-3j\omega) \), for \( 0 \leq \omega \leq \pi/4 \)
   \[ 0 \text{ for } \pi/4 \leq \omega \leq \pi \]
   using a Rectangular window with \( N=7 \). MAY/JUNE 2008
7. (a) Determine the direct form of following system.
   \[ H(z) = 1 + 2z^{-1} - 3z^{-2} + 4z^{-3} - 5z^{-4} \]
   (b) Obtain the cascade form realizations of FIR systems.
   \[ H(z) = 1 + 5/2z^{-1} + 2z^{-2} + 2z^{-3} \]
8. Design a low pass filter using rectangular window by taking 9 samples of \( w(n) \) and with a cutoff frequency of 1.2 radians/sec. Using frequency sampling method, design a band pass FIR filter with the following specification. Sampling frequency \( F_s = 8000 \) Hz, Cutoff frequency \( f_{c1} = 1000 \) Hz, \( f_{c2} = 3000 \) Hz. Determine the filter coefficients for \( N = 7 \). DEC 2009

9. Design and implement linear phase FIR filter of length \( N = 15 \) which has following unit sample sequence \( H(k) = 1 \) ; for \( k = 0, 1, 2, 3 \) 0 ; for \( k = 4, 5, 6, 7 \). NOV/DEC 2008.

10. Explain the method of design of IIR filters using bilinear transform method. MAY/JUNE 2008

11. (a) Discuss the limitations of designing an IIR filter using impulse invariant method
(b) Derive bilinear transformation for an analog filter with system function \( H(s) = b/s + a \).

MAY/JUNE 2007

12. (a) For the analog transfer function
\[
H(s) = \frac{2}{(s+1)(s+3)}
\]
Determine \( H(z) \) using bilinear transformation. With \( T = 0.1 \) sec.
(b) Convert the analog filter
\[
H(s) = \frac{0.5(s+4)}{(s+1)(s+2)}
\]
using impulse invariant transformation \( T = 0.31416 \) s.

NOV/DEC 2008

Unit: 3. DISCRETE FOURIER TRANSFORMS

1. Define DTFT.
The discrete-time Fourier transform (or DTFT) of \( x(n) \) is usually written:
\[
X(\omega) = \sum_{n=-\infty}^{\infty} x[n] e^{-i\omega n}
\]

2. Define Periodicity of DTFT.
Sampling \( x(t) \) causes its spectrum (DTFT) to become periodic. In terms of ordinary frequency \( f \) (cycles per second), the period is the sample rate, \( F_s \). In terms of normalized frequency \( f/F_s \) (cycles per sample), the period is 1. And in terms of \( \omega \) (radians per sample), the period is \( 2\pi \), which also follows directly from the periodicity of \( e^{j\omega n} \). That is:
\[
e^{j(\omega+2\pi k)n} = e^{j\omega n}, \text{ where both } n \text{ and } k \text{ are arbitrary integers. Therefore: } X(\omega+2\pi k) = X(\omega)
\]

3. Difference between DTFT and other transform.
The DFT and the DTFT can be viewed as the logical result of applying the standard continuous Fourier transform to discrete data. From that perspective, we have the satisfying result that it’s not the transform that varies, it’s just the form of the input: If it is
discrete, the Fourier transform becomes a DTFT. If it is periodic, the Fourier transform becomes a Fourier series. If it is both, the Fourier transform becomes a DFT.

4. Write about symmetry property of DTFT.
The Fourier Transform can be decomposed into a real and imaginary or into even and odd.

\[ X(e^{i\omega}) = X_R(e^{i\omega}) + iX_I(e^{i\omega}) \]

or

\[ X(e^{i\omega}) = X_E(e^{i\omega}) + X_O(e^{i\omega}) \]

<table>
<thead>
<tr>
<th>Time Domain</th>
<th>Frequency Domain</th>
</tr>
</thead>
<tbody>
<tr>
<td>(x[n])</td>
<td>(X(e^{i\omega}))</td>
</tr>
<tr>
<td>(x^*[n])</td>
<td>(X^*(e^{-i\omega}))</td>
</tr>
<tr>
<td>(x^*[-n])</td>
<td>(X^*(e^{i\omega}))</td>
</tr>
</tbody>
</table>

5. Define DFT pair.
The sequence of \(N\) complex numbers \(x_0, \ldots, x_{N-1}\) is transformed into the sequence of \(N\) complex numbers \(X_0, \ldots, X_{N-1}\) by the DFT according to the formula:

\[ X_k = \sum_{n=0}^{N-1} x_n e^{-\frac{2\pi i kn}{N}} \quad k = 0, \ldots, N - 1 \]

The inverse discrete Fourier transform (IDFT) is given by

\[ x_n = \frac{1}{N} \sum_{k=0}^{N-1} X_k e^{\frac{2\pi i kn}{N}} \quad n = 0, \ldots, N - 1. \]

6. How will you express IDFT in terms of DFT?
A useful property of the DFT is that the inverse DFT can be easily expressed in terms of the (forward) DFT, via several well-known "tricks". (For example, in computations, it is
often convenient to only implement a fast Fourier transform corresponding to one
transform direction and then to get the other transform direction from the first.)
First, we can compute the inverse DFT by reversing the inputs:
Second, one can also conjugate the inputs and outputs:
Third, a variant of this conjugation trick, which is sometimes preferable because it
requires no modification of the data values, involves swapping real and imaginary parts
(which can be done on a computer simply by modifying pointers). Define \( \text{swap}(x_n) \) as \( x_n \) with its real and imaginary parts swapped—that is, if \( x_n = a + bi \) then \( \text{swap}(x_n) \) is \( b + ai \).

7. Write about Bilateral Z transform.
The bilateral or two-sided Z-transform of a discrete-time signal \( x[n] \) is the function \( X(z) \) defined as
\[
X(z) = \mathcal{Z}\{x[n]\} = \sum_{n=-\infty}^{\infty} x[n]z^{-n}
\]
where \( n \) is an integer and \( z \) is, in general, a complex number:

8. Write about Unilateral Z transform.
Alternatively, in cases where \( x[n] \) is defined only for \( n \geq 0 \), the single-sided or unilateral
Z-transform is defined as
\[
X(z) = \mathcal{Z}\{x[n]\} = \sum_{n=0}^{\infty} x[n]z^{-n}
\]
In signal processing, this definition is used when the signal is causal.

The region of convergence (ROC) is the set of points in the complex plane for which the
Z-transform summation converges.
\[
\text{ROC} = \left\{ z : \left| \sum_{n=-\infty}^{\infty} x[n]z^{-n} \right| < \infty \right\}
\]

10. Write about the output response of Z transform.
If a system \( H(z) \) is driven by a signal \( X(z) \) then the output is \( Y(z) = x(z)H(z) \). By
performing partial fraction decomposition on and then taking the inverse Z-transform the
output can be found. In practice, it is often useful to fractionally decompose before multiplying that quantity by to generate a form of which has terms with easily computable inverse Z-transforms.

11. Define Twiddle Factor.
A twiddle factor, in fast Fourier transform (FFT) algorithms, is any of the trigonometric constant coefficients that are multiplied by the data in the course of the algorithm.

12. State the condition for existence of DTFT?
The conditions are, If \( x(n) \) is absolutely summable then DTFT exist otherwise if \( x(n) \) is not absolutely summable then it should have finite energy for DTFT to exit.

13. 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

14. What is the DTFT of unit sample?
The DTFT of unit sample is 1 for all values of \( \omega \).

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

16. Define circularly even sequence.
A Sequence is said to be circularly even if it is symmetric about the point zero on
17. **Define circularly odd sequence.**
A Sequence is said to be circularly odd if it is anti symmetric about point \( x(0) \) on the circle.

18. **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.

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

20. **State Parseval's theorem.**
Consider the complex valued sequences \( x(n) \) and \( y(n) \). If
\[
x(n) \leftrightarrow X(k)
y(n) \leftrightarrow Y(k)
\]
then \( x(n) \otimes y(n) = \frac{1}{N} X(k)Y^*(k) \)

21. **Find Z transform of \( x(n)=\{1,2,3,4\} \).**
\[
x(n) = \{1,2,3,4\}
\]
\[
X(z) = 1+2z^{-1}+3z^{-2}+4z^{-3}.
\]

22. **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.

23. **State initial value theorem.**
If \( x(n) \) is causal sequence then its initial value is given by \( x(0) = \lim_{z \to \infty} z X(z) \).

27. **List the methods of obtaining inverse Z transform.**
Inverse z transform can be obtained by using
- Partial fraction expansion.
- Contour integration
28. Obtain the inverse z transform of \( X(z) = \frac{1}{z-a}, |z|>|a| \)

Given \( X(z) = \frac{z-1}{1-az^{-1}} \)

By time shifting property \( X(n) = a^n u(n-1) \)

12-MARKS:
1. a) Compute 4-point DFT of casual three sample sequence is given by, \( x(n) = \frac{1}{3}, 0 \leq n \leq 2 \) 0, elsewhere.
   b) State and prove shifting property of DFT. APRIL/MAY 2008
2. Derive and draw the radix -2 DIT algorithms for FFT of 8 points. DEC 2009
3. Compute the DFT for the sequence \( \{1, 2, 0, 0, 2, 1, 1\} \). Using radix -2 DIF FFT and radix -2 DIT-FFT algorithm. OV/DEC 2009
4. Find the output \( y(n) \) of a filter whose impulse response is \( h(n) = \{1, 1, 1\} \) and input signal \( x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\} \). Using Overlap add overlap save method.

5. In an LTI system the input \( x(n) = \{1, 1, 1\} \) and the impulse response \( h(n) = \{-1, 1\} \). Determine the response of LTI system by radix -2 DIT FFT. APRIL/MAY 2007
6. Find the output \( y(n) \) of a filter whose impulse response is \( h(n) = \{1, 1, 1\} \) and input signal \( x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\} \). Using Overlap save method. MAY/JUNE 2006
7. Derive DIF radix 2 FFT algorithm.
8. Derive DIF radix 2 FFT algorithm.
9. Compute the DFT of \( x(n) = a^n \).
10. Determine the circular convolution of the two 4 point sequence \( g(n) = \{1, 2, 0, 1\} \) and \( h(n) = \{2, 2, 1, 1\} \).

Unit : 4.EFFECTS OF FINITE REGISTER LENGTH
1. What do you understand by a fixed-point number?
   In fixed point arithmetic the position of the binary point is fixed. The bits to the right represent the fractional part of the number & those to the left represent the integer part.
   For example; the binary number 01.1100 has the value 1.75 in decimal.

2. What is the objective of spectrum estimation?
The main objective of spectrum estimation is the determination of the power spectral density of a random process. The estimated PSD provides information about the
structure of the random process which can be used for modeling, prediction or filtering of the deserved process.

3. **What is meant by block floating point representation? What are its advantages?**
   In block point arithmetic the set of signals to be handled is divided into blocks. Each block has the same value for the exponent. The arithmetic operations with in the block uses fixed point arithmetic & only one exponent per block is stored thus saving memory. This representation of numbers is more suitable in certain FFT flow graph & in digital audio applications.

4. **What are the advantages of floating point arithmetic?**
   1. Large dynamic range 2. Over flow in floating point representation is unlike.

5. **What are the three-quantization errors to finite word length registers in digital filters?**
   1. Input quantization error 2. Coefficient quantization error 3. Product quantization error

6. **How the multiplication & addition are carried out in floating point arithmetic?**
   In floating point arithmetic, multiplication are carried out as follows,
   Let \( f_1 = M_1 \cdot 2^{c_1} \) and \( f_2 = M_2 \cdot 2^{c_2} \). Then \( f_3 = f_1 \cdot f_2 = (M_1 \cdot M_2) \cdot 2^{(c_1+c_2)} \).
   That is, mantissa is multiplied using fixed-point arithmetic and the exponents are added. The sum of two floating-point numbers is carried out by shifting the bits of the mantissa of the smaller number to the right until the exponents of the two numbers are equal and then adding the mantissas.

7. **What do you understand by input quantization error?**
   In digital signal processing, the continuous time input signals are converted into digital using a b-bit ACD. The representation of continuous signal amplitude by a fixed digit produce an error, which is known as input quantization error.

8. **What is the relationship between truncation error \( e \) and the bits \( b \) for representing a decimal into binary?**
For a 2's complement representation, the error due to truncation for both positive and negative values of $x$ is $-2^b \leq e \leq 2^b$

Where $b$ is the number of bits and $x_t$ is the truncated value of $x$. The equation holds good for both sign magnitude, 1’s complement if $x>0$ if $x<0$, then for sign magnitude and for 1’s complement the truncation error satisfies.

9. What is meant rounding? Discuss its effect on all types of number representation?  
NOV/DEC 2006

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

For fixed point arithmetic, the error made by rounding a number to $b$ bits satisfy the inequality $-2^{b-1}/2 \leq e \leq 2^{b-1}/2$ for all three types of number systems, i.e., 2’s complement, 1’s complement & sign magnitude.

For floating point number the error made by rounding a number to $b$ bits satisfy the inequality $-2^b \leq e \leq 2^b$

10. What is meant by A/D conversion noise? MAY/JUNE 2009

A DSP contains a device, A/D converter that operates on the analog input $x(t)$ to produce $x_q(t)$ which is binary sequence of 0s and 1s.

At first the signal $x(t)$ is sampled at regular intervals to produce a sequence $x(n)$ is of infinite precision. Each sample $x(n)$ is expressed in terms of a finite number of bits given the sequence $x_q(n)$. The difference signal $e(n)=x_q(n)-x(n)$ is called A/D conversion noise.

11. What is the effect of quantization on pole location? (Apr 99)

Quantization of coefficients in digital filters lead to slight changes in their value. This change in value of filter coefficients modifies the pole-zero locations. Some times the pole locations will be changed in such a way that the system may drive into instability.

12. Which realization is less sensitive to the process of quantization?  
Cascade form.
Let us assume a sinusoidal signal varying between +1 and -1 having a dynamic range 2.
If the ADC used to convert the sinusoidal signal employs b+1 bits including sign bit, the number of levels available for quantizing x(n) is 2b+1. Thus the interval between successive levels q = 2\(^{- (b+1)}\).
Where q is known as quantization step size.

14. How would you relate the steady-state noise power due to quantization and the b bits representing the binary sequence?
Steady state noise power
Where b is the number of bits excluding sign bit.

15. What is overflow oscillation? (Oct 98)
The addition of two fixed-point arithmetic numbers cause over flow the sum exceeds the word size available to store the sum. This overflow caused by adder make the filter output to oscillate between maximum amplitude limits. Such limit cycles have been referred to as over flow oscillations.

16. What are the methods used to prevent overflow? NOV/DEC 2009
There are two methods used to prevent overflow
1. Saturation arithmetic
2. Scaling

17. What are the two kinds of limit cycle behavior in DSP?
1. Zero input limit cycle oscillations 2. Overflow limit cycle oscillations

18. What is meant by "dead band" of the filter? NOV/DEC 2009
The limit cycle occur as a result of quantization effect in multiplication. The amplitudes of the output during a limit cycle are confined to a range of values called the dead band of the filter.

19. Explain briefly the need for scaling in the digital filter implementation. (Apr 2000)
To prevent overflow, the signal level at certain points in the digital filter must be scaled so that no overflow occurs in the adder.

20. What is meant by autocorrelation?
The autocorrelation of a sequence is the correlation of a sequence with its shifted version, and this indicates how fast the signal changes.

21. Define noise transfer function (NTF)?
The NTF is defined as the transfer function from the noise source to the filter output. The NTF depends on the structure of the digital network.

22. What are the two types of quantization employed in digital system?
The two types of quantization in digital system are truncation and rounding.

23. What is truncation?
The process of reducing the size of binary number by discarding all bits less significant than the least significant bit that is retained.

24. What is the drawback in saturation arithmetic? APR/MAY 2007
The saturation arithmetic introduces non-linearity in the adder which creates signal distortion.

25. What are the types of arithmetic used in digital computers?
The floating point arithmetic and two’s complement arithmetic are the two types.

12 MARKS:
1. Explain in details about quantization in floating point realizations of IIR filter?
2. Describe the effects of quantization in IIR filter. Consider a first order filter with difference equation y(n) = x(n)+0.5 y(n-1) assume that the data register length is three bits plus a sign bit. The input x(n) = 0.875Z(n). Explain the limit cycle oscillations in the above filter, if quantization is performed by means of rounding and signed magnitude representation is used.MAY/JUNE 2009
3. Explain briefly NOV/DEC 2008
   (i) Effects of coefficient quantization in filter design.
   (ii) Effects of product round off error in filter design.
   (iii) Speech recognition
4. Explain briefly
(i) Define limit cycle oscillation. Explain. (Apr 2000)
(ii) Explain the different representation of fixed and floating point representation.
5. Two first order LPF whose system functions are given below connected in cascade. Determine the over all output noise power $H_1(z) = 1/1 - 0.9z^{-1}$ and $H_2(z) = 1/1 - 0.8z^{-1}$
6. (a) Describe the quantization error occur in rounding and truncation in twos complement.
(b) Draw a sample and hold circuit and explains its operation?
7. (a) Explain dead band in limit cycles? APR/MAY 2007
(b) Draw the statically model of fixed point product quantization and explain
8. (a) What is meant by finite word length effects on digital filters? List them.
(b) Explain fixed point representation of binary numbers. Nov/Dec 2006
9. Explain in detail about binary floating point representation of numbers. May/June 2009
10. Derive the equation for quantization noise power May/June 2009
11. Discuss the round off effects in digital filters. April/May 2010
12. (a) What is dead band of a filter? Derive the dead band of second order linear filter?
(b) Consider all pole second order IIR filter described by equation $y(n) = -0.5y(n-1) - 0.75y(n-2) + x(n)$. Assuming 8 bits to represent pole, determine the dead band region governing the limit cycle. (NOV/DEC 2009)
13. Determine the variance of the round off noise at the output of two cascaded of the filter with system function $H(z) = H_1(z) . H_2(z)$ where $H_1(z) = 1/1 - 0.5z^{-1}$ $H_2(z) = 1/1 - 0.25z^{-1}$
14. Explain with suitable examples the truncation and rounding off errors
15. a) Explain the application of DSP in Speech processing?
   b) What is a vocoder? Explain with a block diagram?
16. Determine the dead band of the filter of $y(n) = 0.95y(n-1) + x(n)$