Digital Signal Processing

Multiple Choice Questions and Answers :-

1. If x(n) is a discrete-time signal, then the value of x(n) at non integer value of ‘n’ is:

a) Zero

b) Positive

c) Negative

d) Not defined

Answer: d

Explanation: For a discrete time signal, the value of x(n) exists only at integral values of n. So, for a non- integer value of ‘n’ the value of x(n) does not exist.

2. The discrete time function defined as u(n)=n for n=0;=0 for n<0 is an:

a) Unit sample signal

b) Unit step signal

c) Unit ramp signal

d) None of the mentioned

Answer: c

Explanation: When we plot the graph for the given function, we get a straight line passing through origin with a unit positive slope. So, the function is called as unit ramp signal.

3.The phase function of a discrete time signal x(n)=an, where a=r.ej? is:

a) tan(n?)

b) n?

c) tan-1(n?)

d) None of the mentioned

Answer: b

Explanation: Given x(n)=an=(r.ej?)n =rn.ejn?

=>x(n)=rn.(cosn?+jsinn?)

Phase function is tan-1(cosn?/sinn?)=tan-1(tan n?)=n?

4. The signal given by the equation4 is known as:

a) Energy signal

b) Power signal

c) Work done signal

d) None of the mentioned

Answer: a

5.Explanation: We have used the magnitude-squared values of x(n), so that our definition applies to

complex-valued as well as real-valued signals. If the energy of the signal is finite i.e., 0<E<8 then

the given signal is known as Energy signal.5. x(n)*d(n-k)=?

a) x(n)

b) x(k)

c) x(k)*d(n-k)

d) x(k)*d(k)

Answer: c

6.Explanation: The given signal is defined only when n=k by the definition of delta function. So, x(n) *d(n-k)= x(k)*d(n-k)

6. A real valued signal x(n) is called as anti-symmetric if:

a) x(n)=x(-n)

b) x(n)=-x(-n)

c) x(n)=-x(n)

d) None of the mentioned

Answer: b

Explanation: According to the definition of anti-symmetric signal, the signal x(n) should be symmetric over origin. So, for the signal x(n) to be symmetric, it should satisfy the condition x(n)=-x(-n).

7. The odd part of a signal x(t) is:

a) x(t)+x(-t)

b) x(t)-x(-t)

c) (1/2)*(x(t)+x(-t))

d) (1/2)*(x(t)-x(-t))

Answer: d

Explanation: Let x(t)=xe(t)+xo(t)

=>x(-t)=xe(-t)-xo(-t)

By subtracting the above two equations, we get

xo(t)=(1/2)*(x(t)-x(-t))

8. Time scaling operation is also known as:

a) Down-sampling

b) Up-sampling

c) Sampling

d) None of the mentioned

Answer: a

Explanation: If the signal x(n) was originally obtained by sampling a signal xa(t), then x(n)=xa(nT). Now, y(n)=x(2n)(say)=xa(2nT). Hence the time scaling operation is equivalent to changing the sampling rate from 1/T to 1/2T, that is to decrease the rate by a factor of 2. So, time scaling is also called as down-sampling.

9. What is the condition for a signal x(n)=Brn where r=eaT to be called as an decaying exponential signal?

a) $0<r<8$

b) $0<r<1$

c) $r>1$

d) $r<0$

Answer: b

Explanation: When the value of ‘r’ lies between 0 and 1 then the value of x(n) goes on decreasing exponentially with increase in value of ‘n’. So, the signal is called as decaying exponential signal.

10. The function given by the equation x(n)=1, for n=0;=0, for n?0 is a:

a) Step function

b) Ramp function

c) Triangular function

d) Impulse function

Answer: d

Explanation: According to the definition of the impulse function, it is defined only at n=0 and is not defined elsewhere which is as per the signal given.

11. Which of the following should be done in order to convert a continuous-time signal to a discrete-time signal?

a) Sampling

b) Differentiating

c) Integrating

d) None of the mentioned

Answer: a

Explanation: The process of converting a continuous-time signal into a discrete-time signal by taking samples of continuous time signal at discrete time instants is known as 'sampling'.

12. The process of converting discrete-time continuous valued signal into discrete-time discrete valued(digital) signal is known as:

a) Sampling

b) Quantization

c) Coding

d) None of the mentioned

Answer: b

Explanation: In this process, the value of each signal sample is represented by a value selected from a finite set of possible values. Hence this process is known as ‘quantization’

13. The difference between the unquantized x(n) and quantized xq(n) is known as:

a) Quantization coefficient

b) Quantization ratio

c) Quantization factor

d) Quantization error

Answer: d

Explanation: Quantization error is the difference in the signal obtained after sampling i.e., x(n) and the signal obtained after quantization i.e., xq(n) at any instant of time.

14. Which of the following is a digital-to-analog conversion process?

a) Staircase approximation

b) Linear interpolation

c) Quadratic interpolation

d) All of the mentioned

Answer: d

Explanation: The process of joining in terms of steps is known as staircase approximation, connecting two samples by a straight line is known as Linear interpolation, connecting three samples by fitting a quadratic curve is called as Quadratic interpolation.

15. The relation between analog frequency ‘F’ and digital frequency ‘f’ is:

a) F=f*T(where T is sampling period)

b) f=F*T

c) No relation

d) None of the mentioned

Answer: b

Explanation: Consider an analog signal of frequency ‘F’, which when sampled periodically at a rate

Fs=1/T samples per second yields a frequency of f=F/Fs=>f=F*T.

16. What is output signal when a signal x(t)=cos(2*pi*40*t) is sampled with a sampling frequency of

20Hz?

a) cos(pi*n)

b) cos(2*pi*n)

c) cos(4*pi*n)

d) cos(8*pi*n)

Answer: c

Explanation: From the question F=40Hz, Fs=20Hz

=>f=F/Fs

=>f=40/20

=>f=2Hz

=>x(n)=cos(4*pi*n)

17. If ‘F’ is the frequency of the analog signal, then what is the minimum sampling rate required to

avoid aliasing?

a) F

b) 2F

c) 3F

d) 4F

Answer: a

Explanation: According to Nyquist rate, to avoid aliasing the sampling frequency should be equal to twice of the analog frequency.

18. What is the nyquist rate of the signal x(t)=3cos(50*pi*t)+10sin(300*pi*t)-cos(100*pi*t)?

a) 50Hz

b) 100Hz

c) 200Hz

d) 300Hz

Answer: d

Explanation: The frequencies present in the given signal are F1=25Hz, F2=150Hz, F3=50Hz

Thus Fmax=150Hz and from the sampling theorem,

nyquist rate=2*Fmax

Therefore, Fs=2*150=300Hz.

19. What is the discrete-time signal obtained after sampling the analog signal x(t)=cos(2000*pi*t)+sin (5000*pi*t) at a sampling rate of 5000samples/sec?

a) cos(2.5*pi*n)+sin(pi*n)

b) cos(0.4*pi*n)+sin(pi*n)

c) cos(2000*pi*n)+sin(5000*pi*n)

d) None of the mentioned

Answer: b

Explanation: From the given analog signal, F1=1000Hz F2=2500Hz and Fs=5000Hz

=>f1=F1/Fs and f2=F2/Fs

=>f1=0.2 and f2=0.5

=>x(n)= cos(0.4*pi*n)+sin(pi*n)

20. If the sampling rate Fs satisfies the sampling theorem, then the relation between quantization

errors of analog signal(eq(t)) and discrete-time signal(eq(n)) is:

a) eq(t)=eq(n)

b) eq(t)eq(n)

d) Not related

Answer: a

Explanation: If it obeys sampling theorem, then the only error in A/D conversion is quantization error.

So, the error is same for both analog and discrete-time signal.

21. The quality of output signal from a A/D converter is measured in terms of:

a) Quantization error

b) Quantization to signal noise ratio

c) Signal to quantization noise ratio

d) Conversion constant

Answer: c

Explanation: The quality is measured by taking the ratio of noises of input signal and the quantized signal i.e., SQNR and is measured in terms of dB.

22. Which bit coder is required to code a signal with 16 levels?

a) 8 bit

b) 4 bit

c) 2 bit

d) 1 bit

Answer: b

Explanation: To code a signal with L number of levels, we require a coder with (log L/log 2) number of bits. So, log16/log2=4 bit coder is required.

23. Which of the following is done to convert a continuous time signal into discrete time signal?

a) Modulating

b) Sampling

c) Differentiating

d) Integrating

Answer: b

Explanation: A discrete time signal can be obtained from a continuous time signal by replacing t by nT, where T is the reciprocal of the sampling rate or time interval between the adjacent values. This procedure is known as sampling.

24. The deflection voltage of an oscilloscope is a ‘deterministic’

signal. True or False?

a) True

b) False

Answer: a

Explanation: The behavior of the signal is known and can be represented by a saw tooth wave form. So,

the signal is deterministic.

25. The even part of a signal x(t) is:

a) x(t)+x(-t)

b) x(t)-x(-t)

c) (1/2)*(x(t)+x(-t))

d) (1/2)*(x(t)-x(-t))

Answer: c

Explanation: Let x(t)=xe(t)+xo(t)

=>x(-t)=xe(-t)-xo(-t)

By adding the above two equations, we get

xe(t)=(1/2)*(x(t)+x(-t))

26. Which of the following is the odd component of the signal x(t)=e(jt)?

a) cost

b) j*sint

c) j*cost

d) sint

Answer: b

Explanation: Let $x(t)=e(jt)$

Now, $xo(t)=(1/2)*(x(t)-x(-t))$

$=(1/2)*(e(jt) - e(-jt))$

$=(1/2)*(\cos t+j\sin t-\cos t+j\sin t)$

$=(1/2)*(2j\sin t)$

$=j*\sin t$

27. For a continuous time signal x(t) to be periodic with a period T, then x(t+mT) should be equal to:

a) x(-t)

b) x(mT)

c) x(mt)

d) x(t)

Answer: d

Explanation: If a signal x(t) is said to be periodic with period T, then x(t+mT)=x(t) for all t and any integer m.

28. Let x1(t) and x2(t) be periodic signals with fundamental periods T1 and T2 respectively. Which of the following must be a rational number for x(t)=x1(t)+x2(t) to be periodic?

a) T1+T2

b) T1-T2

c) T1/T2

d) T1*T2

Answer: c

Explanation: Let T be the period of the signal x(t)

=>x(t+T)=x1(t+mT1)+x2(t+nT2)

Thus, we must have

mT1=nT2=T

=>(T1/T2)=(k/m)= a rational number

29. Let x1(t) and x2(t) be periodic signals with fundamental periods T1 and T2 respectively. Then the

fundamental period of x(t)=x1(t)+x2(t) is:

a) LCM of T1 and T2

b) HCF of T1and T2

c) Product of T1 and T2

d) Ratio of T1 to T2

Answer: a

Explanation: For the sum of x1(t) and x2(t) to be periodic the ratio of their periods should be a

rational number, then the fundamental period is the LCM of T1 and T2.

30. All energy signals will have an average power of:

a) Infinite

b) Zero

c) Positive

d) Cannot be calculated

Answer: b

Explanation: For any energy signal, the average power should be equal to 0 i.e., P=0.

31. x(t) or x(n) is defined to be an energy signal, if and only if the total energy content of the

signal is a:

a) Finite quantity

b) Infinite

c) Zero

d) None of the mentioned

Answer: a

Explanation: The energy signal should have total energy value that lies between 0 and infinity.

32. What is the period of cos2t+sin3t?

a) pi

b) 2*pi

c) 3*pi

d) 4*pi

Answer: b

Explanation: Period of cos2t=(2*pi)/2=pi

Period of sin3t=(2*pi)/3

LCM of pi and (2*pi)/3 is 2*pi.

33. Which of the following justifies the linearity property of z-transform?[x(n)?X(z)] a) x(n)+y(n) ?X

(z)Y(z)

b) x(n)+y(n) ?X(z)+Y(z)

c) x(n)y(n) ?X(z)+Y(z)

d) x(n)y(n) ?X(z)Y(z)

Answer: b

Explanation: According to the linearity property of z-transform, if X(z) and Y(z) are the z-transforms of x(n) and y(n) respectively then, the z-transform of x(n)+y(n) is X(z)+Y(z).

34. What is the z-transform of the signal x(n)=[3(2n)-4(3n)]u(n)?

a) 3/(1-2z-1)-4/(1-3z-1)

b) 3/(1+2z-1)-4/(1+3z-1)

c) 3/(1-2z)-4/(1-3z)

d) None of the mentioned

Answer: a

Explanation: Let us divide the given x(n) into x1(n)= 3(2n)u(n) and x2(n)= 4(3n)u(n) and x(n)=x1(n)-x2 (n)From the definition of z-transform X1(z)= 3/(1-2z-1) and X2(z)= 4/(1-3z-1) So, from the linearity property of z-transform X(z)=X1(z)-X2(z)=> X(z)= 3/(1-2z-1)-4/(1-3z-1)

35. According to Time shifting property of z-transform, if X(z) is the z-transform of x(n) then what is the z-transform of x(n-k)?

a) zkX(z)

b) z-kX(z)

c) X(z-k)

d) X(z+k)

Answer: b

Explanation: According to the definition of Z-transform

36. If X(z) is the z-transform of the signal x(n) then what is the z-transform of anx(n)?

a) X(az)

b) X(az-1)

c) X(a-1z)

d) None of the mentioned

Answer: c

Explanation: We know that from the definition of z-transform

37. If the ROC of X(z) is r1<|z|<r2, then what is the ROC of X(a-1z)?

a) |a|r1<|z|<|a|r2

b) |a|r1>|z|>|a|r2

c) |a|r1<|z|>|a|r2

d) |a|r1>|z|<|a|r2

Answer: a

Explanation: Given ROC of X(z) is r1<|z|<r2

Then ROC of X(a-1z) will be given by r1<|a-1z |<r2=|a|r1<|z|<|a|r2

38. If X(z) is the z-transform of the signal x(n), then what is the z-transform of the signal x(-n)?

a) X(-z)

b) X(z-1)

c) X-1(z)

d) None of the mentioned

Answer: b

Explanation: From the definition of z-transform

39. X(z) is the z-transform of the signal x(n), then what is the z-transform of the signal nx(n)?

a) -z(dX(z))/dz

b) zdX(z)/dz

c) -z-1dX(z)/dz

d) z-1(dX(z))/dz

Answer: a

Explanation:From the definition of z-transform

40. Which of the following relations are true if x(n) is real?

a) X(?)=X(-?)

b) X(?)= -X(-?)

c) X*(?)=X(?)

d) X*(?)=X(-?)

Answer: d

Explanation: We know that, if x(n) is a real sequence

41. If x(n) is a real signal, then 5a

a) True

b) False

Answer: a

Explanation: We know that if x(n) is a real signal, then xI(n)=0 and xR(n)=x(n)

We know that, 5b

Since both XR(?) cos?n and XI(?) sin?n are even, x(n) is also even=> 5a

42. What is the value of XR(?) given X(?)=1/(1-ae-j?) ,|a|<1? a) asin?/(1-2acos?+a2)

b) (1+acos?)/(1-2acos?+a2)

c) (1-acos?)/(1-2acos?+a2)

d) (-asin?)/(1-2acos?+a2)

Answer: c

Explanation: Given, X(?)= 1/(1-ae-j?) ,|a|<1 By multiplying both the numerator and denominator of the above equation by the complex conjugate of the denominator, we obtain X(?)= (1-ae^j?)/((1-ae^(-j?))(1 -ae^j?)) = (1-acos?-jasin?)/(1-2acos?+a^2) This expression can be subdivided into real and imaginary parts, thus we obtain XR(?)= (1-acos?)/(1-2acos?+a2).

43. What is the value of XI(?) given X(?)=1/(1-ae-j?) ,|a|<1? a) asin?/(1-2acos?+a2)

b) (1+acos?)/(1-2acos?+a2)

c) (1-acos?)/(1-2acos?+a2)

d) (-asin?)/(1-2acos?+a2)

Answer: d

Explanation: Given, X(?)= 1/(1-ae-j?) ,|a|<1 By multiplying both the numerator and denominator of the above equation by the complex conjugate of the denominator, we obtain X(?)= (1-ae^j?)/((1-ae^(-j?))(1

-ae^j?)) = (1-acos?-jasin?)/(1-2acos?+a^2) This expression can be subdivided into real and imaginary

parts, thus we obtain XI(?)= (-asin?)/(1-2acos?+a2).

44. What is the value of |X(?)| given X(?)=1/(1-ae-j?) ,|a|<1? a) 1/v(1-2acos?+a2)

b) 1/v(1+2acos?+a2)

c) 1/(1-2acos?+a2)

d) 1/(1+2acos?+a2)

Answer: a

Explanation: For the given X(?)=1/(1-ae-j?) ,|a|<1 we obtain XI(?)= (-asin?)/(1-2acos?+a2) and XR(?)=

(1-acos?)/(1-2acos?+a2)

We know that |X(?)|=v(?X_R (?)?^2+?X_I (?)?^2)

Thus on calculating, we obtain

|X(?)|= 1/v(1-2acos?+a2)

45. What is the Fourier transform of the signal x(n)=a|n|, |a|<1? a) (1+a2)/(1-2acos?+a2)

b) (1-a2)/(1-2acos?+a2)

c) 2a/(1-2acos?+a2)

d) None of the mentioned

Answer: b

Explanation: First we observe x(n) can be expressed as

x(n)=x1(n)+x2(n)

where x1(n)= an, n>0

=0, elsewhere

x2(n)=a-n, n<0 =0, elsewhere Now applying Fourier transform for the above two signals, we get X1(?)=

1/(1-ae^(-j?)) and X2(?)= (ae^j?)/(1-ae^j?) Now, X(?)= X1(?)+ X2(?)= 1/(1-ae^(-j?))+(ae^j?)/(1-ae^j?

) = (1-a2)/(1-2acos?+a2)

46. If X(?) is the Fourier transform of the signal x(n), then what is the Fourier transform of the

signal x(n-k)?

a) ej?k. X(-?)

b) ej?k. X(?)

c) e-j?k. X(-?)

d) e-j?k. X(?)

Answer: d

Explanation: Given

47. What is the convolution of the sequences of x1(n)=x2(n)={1,1,1}?

a) {1,2,3,2,1}

b) {1,2,3,2,1}

c) {1,1,1,1,1}

d) {1,1,1,1,1}

Answer: a

Explanation: Given x1(n)=x2(n)={1,1,1}

By calculating the Fourier transform of the above two signals, we get

X1(?)= X2(?)=1+ ej? + e -j? = 1+2cos?

From the convolution property of Fourier transform we have,

X(?)= X1(?). X2(?)=(1+2cos?)2=3+4cos?+2cos2?

By applying the inverse Fourier transform of the above signal, we get

x1(n)*x2(n)={1,2,3,2,1}

48. What is the energy density spectrum of the signal x(n)=anu(n), |a|<1? a) 1/(1+2acos?+a2)

b) 1/(1-2acos?+a2)

c) 1/(1-2acos?-a2)

d) 1/(1+2acos?-a2)

Answer: b

Explanation: Given x(n)= anu(n), |a|<1 The auto correlation of the above signal is rxx(l)=1/(1-a2) a|

l|, -8< l <8 According to Wiener-Khintchine Theorem, Sxx(?)=F{ rxx(l)}= [1/(1-a2)].F{a|l|} = 1/(1-

2acos?+a2)

49. When the frequency band is selected we can specify the sampling rate and the characteristics of the

pre filter, which is also called as __ filter?

a) Analog filter

b) Anti aliasing filter

c) Both a& b

d) None of the mentioned

Answer: b

Explanation: Once the desired frequency band is selected w e can specify the sampling rate and the

characteristics of the pre filter, which is also called an anti aliasing filter. The anti aliasing filter is an analog filter which has a twofold purpose.

50. What are the main characteristics of Anti aliasing filter?

a) Ensures that bandwidth of signal to be sampled is limited to frequency range.

b) To limit the additive noise spectrum and other interference, which corrupts the signal.

c) Both a& b

d) None of the mentioned

Answer: c

Explanation: T he anti aliasing filter is an analog filter which has a twofold purpose. First, it ensures that the bandwidth of the signal to be sampled is limited to the desired frequency range. Using an antialiasing filter is to limit the additive noise spectrum and other interference, which often corrupts the desired signal. Usually, additive noise is wideband and exceeds the bandwidth of the desired signal.

51. In general, a digital system designer has better control of tolerances in a digital signal processing system than an analog system designer who is designing an equivalent analog system.

a) True

b) False

Answer: a

Explanation: Analog signal processing operations cannot be done very precisely either, since electronic components in analog systems have tolerances and they introduce noise during their operation. In general, a digital system designer has better control of tolerances in a digital signal processing system than an analog system designer who is designing an equivalent analog system.

52. The selection o f the sampling rate Fs=1/T, where T is the sampling interval, not only determines the highest frequency (Fs/2) that is preserved in the analog signal, but also serves as a scale factor that influences the design specifications for digital filters

a) True

b) False

Answer: a

Explanation: Once we have specified the pre filter requirements and have selected the desired sampling rate, w e can proceed with the design of the digital signal processing operations to be performed on the discrete-time signal. The selection of the sampling rate Fs=1/T , where T is the sampling interval, not only determines the highest frequency (Fs/2) that is preserved in the analog signal, but also serves as a scale factor that influences the design specifications for digital filters and any other
discrete-time systems through which the signal is processed.

53. What is the configuration of system for digital processing of an analog signal?

a) Analog signal|| Pre-filter ->D/A Converter -> Digital Processor -> A/D Converter -> Post-filter

b) Analog signal|| Pre-filter ->A/D Converter -> Digital Processor -> D/A Converter -> Post-filter

c) Analog signal|| Post-filter ->D/A Converter -> Digital Processor -> A/D Converter -> Pre-filter

d) None of the mentioned

Answer: b

Explanation: The anti-aliasing filter is an analog filter which has a twofold purpose.

Analog signal|| Pre-filter ->A/D Converter -> Digital Processor -> D/A Converter -> Post-filter

54. In DM, further the two integrators at encode are replaced by one integrator placed before comparator, and then such system is called?

a) System-delta modulation

b) Sigma-delta modulation

c) Source-delta modulation

d) None of the mentioned

Answer: b

Explanation: In DM, Furthermore, the two integrators at the encoder can be replaced by a single integrator placed before the comparator. This system is known as sigma-delta modulation (SDM).

55. In IIR Filter design by the Bilinear Transformation, the Bilinear Transformation is a mapping from

a) Z-plane to S-plane

b) S-plane to Z-plane

c) S-plane to J-plane

d) J-plane to Z-plane

Answer: b

Explanation: From the equation,

56. it is clear that transformation occurs from s-plane to z-plane

2. In Bilinear Transformation, aliasing of frequency components is been avoided.

a) True

b) False

Answer: a

Explanation: The bilinear transformation is a conformal mapping that transforms the j?-axis into the unit circle in the z-plane only once, thus avoiding the aliasing.

57. Is IIR Filter design by Bilinear Transformation is the advanced technique when compared to other design techniques?

a) True

b) False

Answer: True

Explanation: Because in other techniques, only lowpass filters and limited class of bandpass filters are been supported. But this technique overcomes the limitations of other techniques and supports more.

58. In the Bilinear Transformation mapping, which of the following are correct?

a) All points in the LHP of s are mapped inside the unit circle in the z-plane

b) All points in the RHP of s are mapped outside the unit circle in the z-plane

c) Both a & b

d) None of the mentioned

Answer: c

Explanation: The bilinear transformation is a conformal mapping that transforms the j?-axis into the unit circle in the z-plane and all the points are linked as mentioned above.

59. In equation 10if r < 1 then o < 0 and then mapping from s-plane to z-plane occurs in which of the following order? a) LHP in s-plane maps into the inside of the unit circle in the z-plane b) RHP in s-plane maps into the outside of the unit circle in the z-plane c) None of the mentioned d) Both a & b [expand title="View Answer"]Answer: a Explanation: In the above equation, if we substitute the values of r, o then we get mapping in the required way[/expand] 11. In equation 10 if r < 1 then o > 0 and then mapping from s-plane to z-plane occurs in which of the following order?

a) LHP in s-plane maps into the inside of the unit circle in the z-plane

b) RHP in s-plane maps into the outside of the unit circle in the z-plane

c) None of the mentioned

d) Both a & b

Answer: b

Explanation: In the above equation, if we substitute the values of r, o then we get mapping in the required way

60. The lower and upper limits on the convolution sum reflect the causality and finite duration characteristics of the filter.

a) True

b) False

Answer: a

Explanation: We can express the output sequence as the convolution of the unit sample response h(n) of the system with the input signal. The lower and upper limits on the convolution sum reflect the causality and finite duration characteristics of the filter.

61. Which of the following condition should the unit sample response of a FIR filter satisfy to have a linear phase?

a) h(M-1-n) n=0,1,2…M-1

b) ±h(M-1-n) n=0,1,2…M-1

c) -h(M-1-n) n=0,1,2…M-1

d) None of the mentioned

Answer: b

Explanation: An FIR filter has an linear phase if its unit sample response satisfies the condition

h(n)= ±h(M-1-n) n=0,1,2…M-1

62. The roots of the polynomial H(z) are identical to the roots of the polynomial H(z -1).

a) True

b) False

Answer: a

Explanation: We know that 5. This result implies that the roots of the polynomial H(z) are identical to the roots of the polynomial H(z -1).

63. The roots of the equation H(z) must occur in:

a) Identical

b) Zero

c) Reciprocal pairs

d) Conjugate pairs

Answer: c

Explanation: We know that the roots of the polynomial H(z) are identical to the roots of the polynomial

H(z -1). Consequently, the roots of H(z) must occur in reciprocal pairs.

64. If the unit sample response h(n) of the filter is real, complex valued roots need not occur in

complex conjugate pairs.

a) True

b) False

Answer: b

Explanation: We know that the roots of the polynomial H(z) are identical to the roots of the polynomial

H(z -1). This implies that if the unit sample response h(n) of the filter is real, complex valued roots

must occur in complex conjugate pairs.

65. What is the value of h(M-1/2) if the unit sample response is anti-symmetric?

a) 0

b) 1

c) -1

d) None of the mentioned

Answer: a

Explanation: When h(n)=-h(M-1-n), the unit sample response is anti-symmetric. For M odd, the center point of the anti-symmetric is n=M-1/2. Consequently, h(M-1/2)=0.

66. What is the number of filter coefficients that specify the frequency response for h(n) symmetric?
a) (M-1)/2 when M is odd and M/2 when M is even
b) (M-1)/2 when M is even and M/2 when M is odd
c) (M+1)/2 when M is even and M/2 when M is odd
d) (M+1)/2 when M is odd and M/2 when M is even

Answer: d

Explanation: We know that, for a symmetric h(n), the number of filter coefficients that specify the frequency response is (M+1)/2 when M is odd and M/2 when M is even.

67. What is the number of filter coefficients that specify the frequency response for h(n) anti-symmetric?
a) (M-1)/2 when M is even and M/2 when M is odd
b) (M-1)/2 when M is odd and M/2 when M is even
c) (M+1)/2 when M is even and M/2 when M is odd
d) (M+1)/2 when M is odd and M/2 when M is even

Answer: b

Explanation: We know that, for a anti-symmetric h(n) h(M-1/2)=0 and thus the number of filter coefficients that specify the frequency response is (M-1)/2 when M is odd and M/2 when M is even.

68. Which of the following is not suitable either as low pass or a high pass filter?

a) h(n) symmetric and M odd
b) h(n) symmetric and M even
c) h(n) anti-symmetric and M odd
d) h(n) anti-symmetric and M even

Answer: c

Explanation: If h(n)=-h(M-1-n) and M is odd, we get H(0)=0 and H(p)=0. Consequently, this is not suitable as either a low pass filter or a high pass filter.

69. The anti-symmetric condition with M even is not used in the design of which of the following linear-phase FIR filter?
a) Low pass
b) High pass
c) Band pass
d) Bans stop

Answer: a

Explanation: When h(n)=-h(M-1-n) and M is even, we know that H(0)=0. Thus it is not used in the design of a low pass linear phase FIR filter.

70. The anti-symmetric condition is not used in the design of low pass linear phase FIR filter.

a) True

b) False

Answer: a

Explanation: We know that if h(n)=-h(M-1-n) and M is odd, we get H(0)=0 and H(p)=0. Consequently, this is not suitable as either a low pass filter or a high pass filter and when h(n)=-h(M-1-n) and M is even, we know that H(0)=0. Thus it is not used in the design of a low pass linear phase FIR filter. Thus the anti-symmetric condition is not used in the design of low pass linear phase FIR filter.

71. Sampling rate conversion by the rational factor I/D is accomplished by what connection of interpolator and decimator?

a) Parallel

b) Cascade

c) Convolution

d) None of the mentioned

Answer: b

Explanation: A sampling rate conversion by the rational factor I/D is accomplished by cascading an interpolator with a decimator.

72. Which of the following has to be performed in sampling rate conversion by rational factor?

a) Interpolation

b) Decimation

c) Either interpolation or decimation

d) None of the mentioned

Answer: a

Explanation: We emphasize that the importance of performing the interpolation first and decimation second, is to preserve the desired spectral characteristics of x(n).

73. Which of the following operation is performed by the blocks given the figure below?

3

a) Sampling rate conversion by a factor I

b) Sampling rate conversion by a factor D

c) Sampling rate conversion by a factor D/I

d) Sampling rate conversion by a factor I/D

Answer: d

Explanation: In the diagram given, a interpolator is in cascade with a decimator which together performs the action of sampling rate conversion by a factor I/D.

74. The Nth root of unity WN is given as:

a) ej2pN

b) e-j2pN

c) e-j2p/N

d) ej2p/N

Answer: c

Explanation: We know that the Discrete Fourier transform of a signal x(n) is given as

75. Which of the following is true regarding the number of computations requires to compute an N-point DFT?

a) N2 complex multiplications and N(N-1) complex additions

b) N2 complex additions and N(N-1) complex multiplications

c) N2 complex multiplications and N(N+1) complex additions

d) N2 complex additions and N(N+1) complex multiplications

Answer: a

Explanation: The formula for calculating N point DFT is given as5 From the formula given at every step of computing we are performing N complex multiplications and N-1 complex additions. So, in a total to perform N-point DFT we perform N2 complex multiplications and N(N-1) complex additions.

76. What is the DFT of the four point sequence x(n)={0,1,2,3}?

a) {6,-2+2j-2,-2-2j}

b) {6,-2-2j,2,-2+2j}

c) {6,-2+2j,-2,-2-2j}

d) {6,-2-2j,-2,-2+2j}

Answer: c

Explanation: The first step is to determine the matrix W4. By exploiting the periodicity property of W4 and the symmetry property

77. If X(k) is the N point DFT of a sequence whose Fourier series coefficients is given by ck, then

which of the following is true?

a) X(k)=Nck

b) X(k)=ck/N

c) X(k)=N/ck

d) None of the mentioned

Answer: a

Explanation: The Fourier series coefficients are given by the expression

78. What is the DFT of the four point sequence x(n)={0,1,2,3}?

a) {6,-2+2j-2,-2-2j}

b) {6,-2-2j,2,-2+2j}

c) {6,-2-2j,-2,-2+2j}

d) {6,-2+2j,-2,-2-2j}

Answer: d

Explanation: Given x(n)={0,1,2,3}

We know that the 4-point DFT of the above given sequence is given by the expression

79. If W4100=Wx200, then what is the value of x?

a) 2

b) 4

c) 8

d) 16

Answer: c

Explanation: We know that according to the periodicity and symmetry property, 100/4=200/x=>x=8.

80. There is no requirement to process the various signals at different rates commensurate with the corresponding bandwidths of the signals.

a) True

b) False

Answer: b

Explanation: In telecommunication systems that transmit and receive different types of signals, there is a requirement to process the various signals at different rates commensurate with the corresponding bandwidths of the signals.

81. What is the process of converting a signal from a given rate to a different rate?

a) Sampling

b) Normalizing

c) Sampling rate conversion

d) None of the mentioned

Answer: c

Explanation: The process of converting a signal from a given rate to a different rate is known as sampling rate conversion.

82. The systems that employ multiple sampling rates are called multi-rate DSP systems.

a) True

b) False

Answer: a

Explanation: Systems that employ multiple sampling rates in the processing of digital signals are called multi rate digital signal processing systems.

83. Which of the following methods are used in sampling rate conversion of a digital signal?

a) D/A convertor and A/D convertor

b) Performing entirely in digital domain

c) None of the mentioned

d) Both of the mentioned

Answer: d

Explanation: Sampling rate conversion of a digital signal can be accomplished in one of the two general methods. One method is to pass the signal through D/A converter, filter it if necessary, and then to resample the resulting analog signal at the desired rate. The second method is to perform the sampling rate conversion entirely in the digital domain.

84. Which of the following is the advantage of sampling rate conversion by converting the signal into analog signal?

a) Less signal distortion

b) Quantization effects

c) New sampling rate can be arbitrarily selected

d) None of the mentioned

Answer: c

Explanation: One apparent advantage of the given method is that the new sampling rate can be arbitrarily selected and need not have any special relationship with the old sampling rate.

85. Which of the following is the disadvantage of sampling rate conversion by converting the signal into analog signal?

a) Signal distortion

b) Quantization effects

c) New sampling rate can be arbitrarily selected

d) Both a & b

Answer: d

Explanation: The major disadvantage by the given type of conversion is the signal distortion introduced by the D/A converter in the signal reconstruction and by the quantization effects in the A/D conversion.

86. In which of the following, sampling rate conversion are used?

a) Narrow band filters

b) Digital filter banks

c) Quadrature mirror filters

d) All of the mentioned

Answer: d

Explanation: There are several applications of sampling rate conversion in multi rate digital signal processing systems, which include the implementation of narrow band filters, quadrature mirror filters and digital filter banks.

87. Which of the following use quadrature mirror filters?

a) Sub band coding

b) Trans-multiplexer

c) Both of the mentioned

d) None of the mentioned

Answer: c

Explanation: There are many applications where quadrature mirror filters can be used. Some of these applications are sub-band coding, trans-multiplexers and many other applications.

88. The sampling rate conversion can be as shown in the figure below.

a) True

b) False

Answer: a

Explanation: The process of sampling rate conversion in the digital domain can be viewed as a linear filtering operation as illustrated in the given figure.

89. If Fx and Fy are the sampling rates of the input and output signals respectively, then what is the value of Fy/Fx?

a) D/I

b) I/D

c) I.D

d) None of the mentioned

Answer: b

Explanation: The input signal x(n) is characterized by the sampling rate Fx and he output signal y(m)

is characterized by the sampling rate Fy, then

Fy/Fx= I/D

where I and D are relatively prime integers.

90. What is the process of reducing the sampling rate by a factor D?

a) Sampling rate conversion

b) Interpolation

c) Decimation

d) None of the mentioned

Answer: c

Explanation: The process of reducing the sampling rate by a factor D, i.e., down-sampling by D is

called as decimation.

91. What is the process of increasing the sampling rate by a factor I?

a) Sampling rate conversion

b) Interpolation

c) Decimation

d) None of the mentioned

Answer: b

Explanation: The process of increasing the sampling rate by a integer factor I, i.e., up-sampling by I is called as interpolation.

92. The reconstruction o f the signal from its samples as a linear filtering process in which a discrete-time sequence of short pulses (ideally impulses) with amplitudes equal to the signal samples, excites an analog filter.

a) True

b) False

Answer: a

Explanation: The reconstruction o f the signal from its samples as a linear filtering process in which a discrete-time sequence of short pulses (ideally impulses) with amplitudes equal to the signal samples, excites an analog filter.

93. The ideal reconstruction filter is an ideal low pass filter and its impulse response extends for all time.

a) True

b) False

Answer: a

Explanation: The ideal reconstruction filter is an ideal low pass filter and its impulse response extends for all time. Hence the filter is noncausal and physically nonrealizable. Although the interpolation filter with impulse response given can be approximated closely with some delay, the resulting function is still impractical for most applications where D /A conversion are required.

94. D /A conversion is usually performed by combining a D /A converter with a sample-and-hold (S/H) and followed by a low pass (smoothing) filter.

a) True

b) False

Answer: a

Explanation: D /A conversion is usually performed by combining a D /A converter with a sample-and hold (S/H) and followed by a low pass (smoothing) filter. T he D /A converter accepts at its input, electrical signals that correspond to a binary word, and produces an output voltage or current that is proportional to the value o f the binary word.

95. The time required for the output o f the D /A converter to reach and remain within a given fraction of the final value, after application of the input code word is called?

a) Converting time

b) Setting time

c) Both a& b

d) None of the mentioned

Answer: b

Explanation: An important parameter o f a D /A converter is its settling time, which is defined as the time required for the output o f the D /A converter to reach and remain within a given fraction (usually,±1/2 LSB) of the final value, after application of the input code word.

96. In D/A converter, the application of the input code word results in a high-amplitude transient, called?

a) Glitch

b) Deglitch

c) Glitter

d) None of the mentioned

Answer: a

Explanation: The application o f the input code word results in a high-amplitude transient, called a "glitch." This is especially the case when two consecutive code words to the A /D differ by several bits.

97. In a D/A converter, the usual way to solve the glitch is to use deglitcher. How is the Deglitcher designed?

a) By using a low pass filter

b) By using a S/H circuit

c) Both a& b

d) None of the mentione

Answer: b

Explanation: The usual way to remedy this problem is to use an S/H circuit designed to serve as a "deglitcher". Hence the basic task of the S/H is to hold the output of the D /A converter constant at the previous output value until the new sample at the output o f the D /A reaches steady state, and then it samples and holds the new value in the next sampling interval. Thus the S/H approximates the analog signal by a series of rectangular pulses whose height is equal to the corresponding value of the signal pulse.