INSTITUTE OF AERONAUTICAL ENGINEERING

(Autonomous)

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ELECTRONICS AND COMMUNICATION ENGINEERING

DEFINITIONS AND TERMINOLOGY QUESTION BANK

Course Name	:	**DIGITAL SIGNAL PROCESSING**
Course Code	:	**AEC012**
Program	:	**B.Tech**
Semester	:	**VI**
Branch	:	**Electronics and Communication Engineering**
Section	:	**A,B,C , D and E**
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OBJECTIVES:

I	To help students to consider in depth the terminology and nomenclature used in the syllabus.
II	To focus on the meaning of new words / terminology/nomenclature

DEFINITIONS AND TERMINOLOGY QUESTION BANK

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
	UNIT-I					
1	Define a signal.	A signal is defined as a single-valued function of one or more independent variables which contain some information.	Understand	CO1	CLO1	AEC012.01
2	What is one dimensional signal?	A signal which depends on only one independent variable is called a one dimensional signal.	Remember	CO1	CLO2	AEC012.02
3	What is signal modeling?	The representation of a signal by the mathematical expression is known as signal modeling.	Understand	CO1	CLO3	AEC012.03
4	Define non-linear system	A system which does not satisfy superposition principle is known as non-linear system	Remember	CO1	CLO1	AEC012.01
5	Define causal 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.	Understand	CO1	CLO2	AEC012.02
6	Define impulse response.	The response of system for a unit impulse input is called impulse response.	Remember	CO1	CLO3	AEC012.03

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
7	What is a discrete time signal?	A discrete time signal x (n) is a function of an independent variable that is an integer. a discrete time signal is not defined at instant between two successive samples.	Remember	CO1	CLO1	AEC012.01
8	What is stability test?	If a system is BIBO stable, then the output will be bounded for every input to the system that is bounded. A signal is bounded if there is a finite value such that the signal magnitude never exceeds , that is for discrete-time signals, or. for continuous-time signals.	Understand	CO1	CLO2	AEC012.02
9	Define Region of Convergence.	The region of convergence (ROC) of X(Z) the set of all values of Z for which X(Z) attain final value.	Remember	CO1	CLO3	AEC012.03
10	What is Linear time invariant system (LTI)?	A LTI system is one which obeys properties of linearity and time invariant. These systems are characterized by unit sample response. Output of LTI system is computed by convolution of input and unit sample response.	Understand	CO1	CLO1	AEC012.01
11	What is a discrete time system?	A discrete time system is an algorithm that performs some prescribed operation on a discrete time signal is called discrete time system.	Understand	CO1	CLO1	AEC012.01
12	Define periodic and aperiodic signal.	A signal x(n) is periodic in period N, if x(n+N) =x(n) for all n. If a signal does not satisfy this equation, the signal is called aperiodic signal.	Remember	CO1	CLO2	AEC012.02
13	What do you mean by fundamental period of a signal?	The smallest value of N that satisfies the condition x(n+N) =x(n) for all values of n for discrete time signal is called the fundamental period of the signal x(n).	Understand	CO1	CLO3	AEC012.03
14	What is an energy signal?	An energy signal is one whose total energy E=finite value and whose average power P=0.	Remember	CO1	CLO1	AEC012.01
15	What are digital signals?	The signals that are discrete in time and quantized in amplitude are called digital signals.	Understand	CO1	CLO1	AEC012.01
16	Define recursive system?	Present output of system depends on present and past input s and also past outputs then the system said to be recursive system.	Understand	CO1	CLO2	AEC012.02
17	What is a odd signal?	A real value signal x(n) is called anti-symmetric (odd) if x(-n) = -x(n).	Remember	CO1	CLO3	AEC012.03

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
18	Define static system.	A discrete time system is called static or memory less if its output at any instant depends almost on the input sample at the same time but not on past and future samples of the input.	Understand	CO1	CLO1	AEC012.01
19	Define dynamic system.	A discrete time system is called dynamic or with memory if its output at any instant depends on the input sample at the same time but not on past and future samples of the input.	Understand	CO1	CLO1	AEC012.01
20	Define time invariant system.	A system is called time invariant if its output, input characteristics do not change with time.	Remember	CO1	CLO2	AEC012.02
21	What is DSP?	Digital signal processing (DSP) refers to performing operations on discrete and digital signals. i.e Processing of discrete signals by means of digital systems.	Remember	CO1	CLO3	AEC012.03
22	Define non recursive system.	If the present output of system depends on present and past inputs only, then the system said to be non - recursive.	Understand	CO1	CLO1	AEC012.01
23	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 combined later to get the output.	Understand	CO1	CLO1	AEC012.01
24	Define digital frequency.	Digital frequency is a unit measurement of frequency equivalent to number of cycles or samples per second.	Understand	CO1	CLO2	AEC012.02
25	What is convolution?	Convolution is a mathematical way of combining two signals to form a third signal. It is the most important technique in digital signal processing, using the strategy of impulse decomposition.	Understand	CO1	CLO3	AEC012.03
26	Why convolution is required?	Convolution is required to determine the response of an LTI system for a given input and impulse response.	Understand	CO1	CLO1	AEC012.01
27	What are the methods of evaluating convolution?	The important methods to evaluate convolution are Overlap-add and over-lap save methods.	Remember	CO1	CLO2	AEC012.02

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
28	What is natural response?	Natural response is system's response to initial conditions with all external forces set to zero.	Understand	CO1	CLO3	AEC012.03
29	Define FIR system.	If the impulse of the system is finite duration then the system is called finite impulse response (FIR) system.	Understand	CO1	CLO1	AEC012.01
30	Define anti-causal signal.	A discrete time signal x(n) is said to be anti-causal if x(n)=0 for n>0.	Remember	CO1	CLO1	AEC012.01
31	Define time variant system.	A system is called time variant if its input, output characteristics changes with time.	Remember	CO1	CLO2	AEC012.02
32	What is a linear system?	Linear system is one which satisfies superposition principle.	Understand	CO1	CLO3	AEC012.03
33	What is a power signal?	A power signal is one whose average power P= finite value and whose total energy E=∞.	Understand	CO1	CLO1	AEC012.02
34	Define even signal.	A real value signal x(n) is called symmetric (even) if x(-n) =x(n).	Remember	CO1	CLO2	AEC012.02
35	What is a non-causal system?	The system is said to be non-causal if the output of the system at any time 'n' also depends future inputs.	Remember	CO1	CLO3	AEC012.03
36	What is cascading?	If the output of one stage is given as input to its next stage then it is called cascading	Remember	CO1	CLO1	AEC012.01
37	What is signal processing?	Signal processing means, changing the basic nature of signal to obtain the desired shaping of the input signal	understand	CO1	CLO2	AEC012.02
38	Define multi channel signal.	Signals are generated by multiple sources or multiple sensors. These signals are represented by vector form. Vector of signals as a multichannel signals.	Remember	CO1	CLO3	AEC012.03
39	Define multi dimensional signal.	A signal is a function of more than two independent variables such type of signals is referred to as multi dimensional signal. A Signal is a function.	Remember	CO1	CLO1	AEC012.01
40	Define 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.	Understand	CO1	CLO2	AEC012.02
41	Define 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	Understand	CO1	CLO1	AEC012.01

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		linear filter. Zero padding is not necessary to find the response of a linear filter.				
42	What is homogeneity property?	Homogeneity property means a system which produces an output y(n).	Understand	CO1	CLO2	AEC012.02
43	What is superposition property?	Superposition property means a system which produces an output $y_1(n)$ for an input $x_1(n)$ and an output $y_2(n)$ for an input $x_2(n)$ must produce $x_1(n)+ x_2(n)$	Remember	CO1	CLO3	AEC012.03
44	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.	Remember	CO1	CLO1	AEC012.01
45	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)*h(n). This process is repeated for all sections and the filtered sections are abutted together.	Remember	CO1	CLO2	AEC012.02
46	Define IIR system.	If the impulse of the system is infinite duration then the system is called infinite impulse response (IIR) system.	Understand	CO1	CLO1	AEC012.01
47	Define impulse response.	The response of system for a unit step input is called step response.	Remember	CO1	CLO2	AEC012.02
48	Define a system.	A system is defined as a physical device, that generates a response or output signal for a given input signal.	Understand	CO1	CLO3	AEC012.03
49	What is an invertible system?	An invertible system is a system which has a unique relation between its input and output.	Understand	CO1	CLO1	AEC012.01
50	What is non-	A non-invertible system is a system which does not have a	Remember	CO1	CLO2	AEC012.02

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
	invertible system?	unique relation between its input and output.				
		UNIT-II				
1	What is zero padding?	Number of zeros adding to given length sequence To obtained desired length of sequence from given length of sequence	Understand	CO2	CLO4	AEC012.04
2	What are uses of zero padding?	To get better display of frequency spectrum .With zero padding, DFT can be used in linear filtering.	Remember	CO2	CLO5	AEC012.05
3	What is meant by in place in DIT and DFT algorithm?	An algorithm that uses the same location to store both the input and output sequence is called in-place algorithm.	Understand	CO2	CLO6	AEC012.06
4	What is Parseval's Theorm?	Parseval's theorem usually refers to the result that the Fourier transform is unitary; that the sum of the square of a function is equal to the sum of the square of its transform.	Understand	CO2	CLO4	AEC012.04
5	Define 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.	Remember	CO2	CLO5	AEC012.05
6	Differentiate linear and circular convolution	Linear convolution is the basic operation to calculate the output for any linear time invariant system given its input and its impulse response. Circular convolution is the same thing but considering that the support of the signal is periodic	Remember	CO2	CLO6	AEC012.06
7	What is meant by sectioned convolution?	If the input data sequence of longer duration, it is very difficult to obtained the output sequence due to limited memory of digital system therefore input data sequence is divided into smaller sections . These sections are processed separately one at a time and combined later to get output.	Understand	CO2	CLO4	AEC012.04
8	List the methods used in sectional convolution.	Over- lap save method and over-lap add method	Remember	CO2	CLO4	AEC012.04
9	Distinguish between DFT and DTFT.	DFT is obtained by performing sampling operation in both time and frequency domain whereas DTFT is obtained by performing sampling operation in time domain only. DFT has discrete frequency spectrum and DTFT has continuous frequency spectrum.	Remember	CO2	CLO5	AEC012.05
10	Why FFT is needed?	Direct computation of N- point DFT requires N^2 complex multiplications and N(N-1) complex additions. Thus for large	Remember	CO2	CLO6	AEC012.06

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		value of N direct evaluation of DFT an unreasonable amount of computation. By using FFT algorithm number of computations can be reduced.				
11	When the DFT $X(k)$ of sequence $x(n)$ is imaginary?	If the sequence $x(n)$ is real and odd or imaginary and even , then $X(k)$ is purely imaginary.	Remember	CO2	CLO4	AEC012.04
12	What is the foremost condition of existence of DTFT for an aperiodic sequence?	That is if sequence is absolutely summable then DTFT exists for that sequence	Remember	CO2	CLO4	AEC012.04
13	What is the Wiener Khint chine theorm?	Wiener–Khinchin–Einstein or Khinchin–Kolmogorov theorem, states that the autocorrelation function of a wide-sense-stationary random process has a spectral decomposition given by the power spectrum of that process	Remember	CO2	CLO5	AEC012.05
14	When the DFT $X(k)$ of sequence $x(n)$ is real?	If the sequence $x(n)$ is real and even or imaginary and odd, then $X(k)$ is purely real.	Remember	CO2	CLO6	AEC012.06
15	What is DTFT?	DTFT is obtained by performing sampling operation in time domain only. DTFT has continuous frequency spectrum for discrete time signal as input	Understand	CO2	CLO4	AEC012.04
16	What is IDTFT?	$x(n)=\frac{1}{2\pi}\int_{-\pi}^{\pi}X(\omega)e^{j\omega n}d\omega$ it transforms continuous frequency spectrum signal into discrete time signal	Understand	CO2	CLO5	AEC012.05
17	What is IDFT?	$x(n)=\frac{1}{N}\sum_{k=0}^{N-1}X(k)e^{j\frac{j2\pi nk}{N}}$ it transforms discrete frequency spectrum to discrete time signal.	Understand	CO2	CLO6	AEC012.06
18	What is DFT?	DFT is obtained by performing sampling operation in both time and frequency domain. DFT has discrete frequency spectrum for discrete time signal as input	Understand	CO2	CLO4	AEC012.04
19	What is periodicity?	The term periodic properties in elements, refers to the properties that recur at regular intervals. The trend of recurrence of properties is called periodicity.	Understand	CO2	CLO5	AEC012.05

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
20	Give the relationship between Fourier series coefficients and discrete Fourier transform.	DFT of given sequence = multiplication of length of finite duration sequence and Fourier series coefficients	Remember	CO2	CLO4	AEC012.04
21	What is the basic operation in DIT algorithm?	The basic operation is butter fly in which two inputs $X_m(p)$ and $X_m(q)$ are combine to give outputs $X_{m+1}(p)$ and $X_{m+1}(q)$ $X_{m+1}(p) = X_m(p) + W_N^K X_m(q)$ $X_{m+1}(q) = X_m(p) - W_N^K X_m(q)$ W_N^K is twiddle factor.	Remember	CO2	CLO5	AEC012.05
22	What is dividing and conquer approach?	This approach is based on the decomposition of an N point DFT into successively smaller DFTs this leads to a family computationally efficient algorithm. N factorized as two integers.	Understand	CO2	CLO6	AEC012.06
23	How Many Multiplications And Additions Are Required To Compute N Point DFT Using Radix-2 FFT?	The number of multiplications and additions required to compute N point DFT using radix-2 FFT are N/2 $\log_2^N$ and N $\log_2^N$ respectively.	Remember	CO2	CLO4	AEC012.04
24	What are properties of twiddle factor?	symmetry : $W_N^{K+\frac{N}{2}} = -W_N^K$ periodicity $W_N^{K+N} = W_N^K$	Remember	CO2	CLO5	AEC012.05
25	Define twiddle factor.	The twiddle factor is given by $e^{-j2\pi/N} = W_N$	Remember	CO2	CLO6	AEC012.06
26	How many complex multiplications and additions are involved in butterfly computation?	One complex multiplication and two complex additions are required in butterfly computation.	Remember	CO2	CLO4	AEC012.04
27	How many butterflies is possible In N point radix-2 FFT?	The number of butterflies possible in radix-2 FFT is N/2.	Remember	CO2	CLO5	AEC012.05
28	What is 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. This remains the term's most common meaning, but it may	Understand	CO2	CLO4	AEC012.04

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		also be used for any data-independent multiplicative constant in an FFT.				
29	What do you understand circular shifting operation?	Given finite duration convert into periodic sequence and apply linear shifting operation and then extract one period of periodic sequence this is called as circular shifting	Understand	CO2	CLO5	AEC012.05
30	What is Parseval's theorem?	Parseval's theorem expresses the energy of finite duration sequence in terms energy of frequency components.	Understand	CO2	CLO6	AEC012.06
31	Define linear convolution.	A mathematical approach to determine the response of an LTI system with input and impulse response. 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.	Remember	CO2	CLO4	AEC012.04
32	What is meant by radix -2?	FFT is most efficient algorithm to compute DFT. if number of output points N can be expressed as power of 2 then this algorithm known as radix -2 FFT algorithms.	Understand	CO2	CLO4	AEC012.04
33	What is FFT?	FFT (fast Fourier transform) is the algorithm to compute DFT with reduced number of complex operations.	Understand	CO2	CLO5	AEC012.05
34	What is speed improvement factor for calculating 64 point DFT using direct computation and FFT algorithm?	Complex multiplications using direct computation of DFT is $64^2 = 4096$. Using FFT number of complex multiplications is $\frac{N}{2}\log_2 N = \frac{64}{2}\log_2 64 = 192$.	Understand	CO2	CLO6	AEC012.06
35	What are the results of DFT of multiplication of two sequences time domain?	DFT of multiplication of two sequences in time domain results circular convolution in frequency domain.	Remember	CO2	CLO4	AEC012.04
36	What are the results of DFT of correlation of two sequences time domain?	DFT of correlation of two sequences in time domain results product of DFT of 1st sequence and complex conjugate of DFT of 2nd sequence,	Understand	CO2	CLO5	AEC012.05
37	What is decimation in time radix -2 FFT algorithms?	This algorithm is based upon that of decomposing the computation of DFT of sequence length N into successively a\smaller and smaller DFTs.	Understand	CO2	CLO4	AEC012.04

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
38	What is decimation in frequency radix - 2 FFT algorithms?	This algorithm is based upon that of decomposing the computation of DFT of sequence length N into successively a\smaller and smaller DFTs.	Understand	CO2	CLO5	AEC012.05
39	What is DTFT?	The Discrete Time Fourier Transform (DTFT) is the member of the Fourier transform family that operates on aperiodic, discrete signals. The best way to understand the DTFT is how it relates to the DFT. To start, imagine that you acquire an N sample signal, and want to find its frequency spectrum.	Remember	CO2	CLO6	AEC012.06
40	What is DFT?	As the name implies, the Discrete Fourier Transform (DFT) is purely discrete: discrete-time data sets are converted into a discrete-frequency representation. This is in contrast to the DTFT that uses discrete time, but converts to continuous frequency.	Understand	CO2	CLO4	AEC012.04
41	What are the properties of DFT	Periodicity Linearity Time reversal Circular time shift Circular frequency shift Complex conjugate Parseval's theorm	Remember	CO2	CLO4	AEC012.04
42	Why DFT is required	Like continuous time signal Fourier transform, discrete time Fourier Transform can be used to represent a discrete sequence into its equivalent frequency domain representation and LTI discrete time system and develop various computational algorithms.			CLO5	AEC012.05
43	Differentiate DFT and DTFT	In DFT, your input signal is the output of your DTFT which is a continuous, periodic frequency domain signal, and DFT gives you the Discrete samples of continuous DTFT. The discrete-time Fourier transform (DTFT) is the (conventional) Fourier transform of a discrete-time signal	Remember	CO2	CLO6	AEC012.06
44	What are the methods to compute DFT	Circular convolution Formula based method	Understand	CO2	CLO4	AEC012.04
45	What is twiddle factor	A twiddle factor, in fast Fourier transform (FFT) algorithms, is any of the trigonometric constant coefficients that are	Remember	CO2	CLO5	AEC012.05

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		multiplied by the data in the course of the algorithm. ... This remains the term's most common meaning, but it may also be used for any data-independent multiplicative constant in an FFT.				
46	Differentiate linear and circular convolution	Linear convolution is the basic operation to calculate the output for any linear time invariant system given its input and its impulse response. Circular convolution is the same thing but considering that the support of the signal is periodic (as in a circle, hence the name)	Remember	CO2	CLO6	AEC012.06
47	What is the foremost condition of existence of DTFT for an aperiodic sequence	$\sum_{n=-\infty}^{\infty} \lvert x(n) < \infty \rvert$ That is if sequence is absolutely summable then DTFT exists for that sequence	Understand	CO2	CLO4	AEC012.04
48	What is Fourier transform of discrete unit impulse signal	$\frac{1}{1-e^{-jw}}$	Remember	CO2	CLO5	AEC012.05
49	When the DFT $X(k)$ of sequence $x(n)$ is imaginary?	If the sequence $x(n)$ is real and odd or imaginary and even , then $X(k)$ is purely imaginary.	Remember	CO2	CLO6	AEC012.06
50	What is the condition of existence of Discrete Time FT for an aperiodic sequence?	Summation of x(n) less than infinite That is if sequence is absolutely summable then DTFT exists for that sequence	Remember	CO2		AEC012.07
UNIT-III						
1	What is impulse invariant mapping?	The philosophy of this technique is to transform an analog prototype filter into an IIR discrete time filter whose impulse response [h(n)] is a sampled version of the analog filter’s impulse response, multiplied by T.	Understand	CO3	CLO7	AEC012.07
2	What is the bilinear transformation?	The bilinear transform is a method of compressing the infinite, straight analog frequency axis to a finite one long enough to wrap around the unit circle only once.	Understand	CO3	CLO8	AEC012.08
3	What are the properties of chebyshev filter?	The magnitude response of the chebyshev filter exhibits ripple either in the stop band or the pass band. The poles of this filter lies on the ellipse	Understand	CO3	CLO9	AEC012.09
4	Define the zeros.	The zeros of the system H(z) are the values of z for which H(z) = 0.	Remember	CO3	CLO10	AEC012.10

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
5	Define the poles.	The poles of the system H(z) are the values of z for which H(z) = ∞.	Understand	CO3	CLO11	AEC012.11
6	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.	Remember	CO3	CLO11	AEC012.11
7	Why feedback is required in IIR systems?	It is required to generate infinitely long impulse response in IIR systems.	Understand	CO3	CLO7	AEC012.07
8	State the structure of IIR filter?	IIR filters are of recursive type whereby the present o/p sample depends on present i/p, past i/p samples and o/p samples. The design of IIR filter is realizable and stable. The impulse response h(n) for a realizable filter is h(n)=0 for n≤0	Remember	CO3	CLO8	AEC012.08
9	What do you understand by backward difference?	It is one of the simplest methods of converting analog to digital filter is to approximate the differential equation by an equivalent difference equation.	Remember	CO3	CLO9	AEC012.09
10	How one can design digital filters from analog filters?	Map the desired digital filter specifications into those for an equivalent analog filter. Derive the analog transfer function for the analog prototype. Transform the transfer function of the analog prototype into an equivalent digital filter transfer function.	Remember	CO3	CLO10	AEC012.10
11	Define procedures for digitizing the transfer function of an analog filter.	The two important procedures for digitizing the transfer function of an analog filter are Impulse invariance method and Bilinear transformation method.	Understand	CO3	CLO7	AEC012.07
12	What do you understand by backward difference?	One of the simplest method for converting an analog filter into a digital filter is to approximate the differential equation by an equivalent difference equation. d/dt y(t)=y(nT)-y(nT-T)/T The above equation is called backward difference equation.	Understand	CO3	CLO8	AEC012.08
13	What is bilinear transformation?	The bilinear transformation is a mapping that transforms the left half of S-plane into the unit circle in the Z-plane only once, thus avoiding aliasing of frequency components. The mapping from the S-plane to the Z-plane is in bilinear transformation is S=2/T(1-Z-1/1+Z-1)	Remember	CO3	CLO9	AEC012.09

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
14	Define the Structure of IIR filter.	IIR filters are of recursive type whereby the present o/p sample depends on present i/p, past i/p samples and o/p samples.	Remember	CO3	CLO10	AEC012.10
15	What is warping effect	The effect of the non-linear compression at high frequencies is called warping effect.	Understand	CO3	CLO10	AEC012.10
16	What is the mapping of bilinear transformation?	The mapping for the bilinear transformation is a one-to-one mapping that is for every point Z, there is exactly one corresponding point S,	Understand	CO3	CLO11	AEC012.11
17	Define pre-warping.	The effect of the non-linear compression at high frequencies can be compensated. this compression can be compensated by introducing a suitable pre-scaling, or pre-warping the critical frequencies by using the formula.	Understand	CO3	CLO7	AEC012.07
18	What is the transposition theorem & transposed structure?	According to transposition theorem if we reverse the directions of all branch transmittance and interchange the input and output in the flow graph, the system function remains unchanged.	Remember	CO3	CLO8	AEC012.08
19	What is transposed structure?	The transpose of a structure is defined by the following operations. Reverse the directions of all branches in the signal flow graph	Understand	CO3	CLO9	AEC012.09
20	Define signal flow graph.	A signal flow graph is a graphical representation of the relationships between the variables of a set of linear difference equations.	Understand	CO3	CLO10	AEC012.10
21	Define signal flow graph.	A signal flow graph is a graphical representation of the relationships between the variables of a set of linear difference equations.	Understand	CO3	CLO10	AEC012.10
22	What are the different arithmetic types in digital systems?	There are three arithmetic types used in digital systems. They are fixed point arithmetic, floating point ,block floating point arithmetic.	Understand	CO3	CLO11	AEC012.11
23	What is meant by fixed point number?	In fixed point number the position of a binary point is fixed. The bit to the right represent the fractional part and those to the left is integer part.	Remember	CO3	CLO7	AEC012.07
24	What are the different IIR Filter types?	Butterworth filter, Chebyshev, Elliptic and Bessel type IIR filters.	Remember	CO3	CLO8	AEC012.08
25	What are the important features of IIR filter?	No linear phase and desired magnitude response characteristics are the important	Remember	CO3	CLO9	AEC012.09

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		features in IIR filter design specifications				
26	What is the main objective of Impulse invariance method?	The main objective of impulse invariant method is to develop an IIR filter whose impulse response is sampled version of analog filter.	Remember	CO3	CLO10	AEC012.10
27	What is the importance of poles in filter design?	The location of the poles decides the stability of the filter and the order of the filter is also given by the number of poles	Understand	CO3	CLO7	AEC012.07
28	What are reasons behind to design digital IIR filter from Analog filter?	Analog filter design is highly advanced since useful results can be achieved. It is advantages to utilize the design procedure already developed for analog filter.	Remember	CO3	CLO8	AEC012.08
29	What is aliasing?	The phenomenon of high frequency components acquiring the identity of low frequency component is called aliasing.	Understand	CO3	CLO9	AEC012.09
30	What is Nyquist sampling rate?	It is the minimum sampling rate required to achieve the sampled signal without aliasing.	Understand	CO3	CLO10	AEC012.10
31	Define Recursive Impulse Response filter.	Based on impulse response the filters are of two The IIR filters are of recursive type, whereby the present output sample depends on the present input, past input samples and output samples.	Understand	CO3	CLO11	AEC012.11
32	What is IIR property?	IIR is a property applying to many linear time-invariant systems. linear time-invariant systems are most electronic and digital filters. Systems with this property are known as IIR systems	Understand	CO3	CLO10	AEC012.10
33	Define IIR system.	If the impulse of the system is infinite duration then the system is called infinite impulse response (IIR) system.	Understand	CO3	CLO7	AEC012.07
34	What is a filter?	Filter is system which removes unwanted frequencies from the input signal	Remember	CO3	CLO8	AEC012.08
35	Define IIR system.	If the impulse of the system is infinite duration then the system is called infinite impulse response (IIR) system.	Understand	CO3	CLO9	AEC012.09
36	What are the steps involved to design digital filter?	The specification of desired properties of system. The approximation of these specifications using a causal discrete time system. The	Remember	CO3	CLO10	AEC012.10

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		realization of system using finite precision arithmetic				
37	Define design a digital IIR – BTM.	Pre-warping the digital specifications. Design an analog filter to meet the pre-warped specifications. Apply the bilinear transformation method.	Remember	CO3	CLO11	AEC012.11
38	How one can design digital filters from analog filters?	Map the desired digital filter specifications into those for an equivalent analog filter. Derive the analog transfer function for the analog prototype.	Understand	CO3	CLO7	AEC012.07
39	How one can design digital filters from analog filters?	Map the desired digital filter specifications into those for an equivalent analog filter. Derive the analog transfer function for the analog prototype. Transform the transfer function of the analog prototype into an equivalent digital filter transfer function.	Understand	CO3	CLO8	AEC012.08
40	What are the design techniques of designing FIR filters?	There are three well known methods for designing FIR filters with linear phase. Window method .Frequency sampling method Optimal or minimax design.	Remember	CO3	CLO9	AEC012.09
41	Define Gibb's phenomenon.	One possible way of finding an FIR filter that approximates H(ejw) would be to truncate the infinite Fourier series at n. Direct truncation of the series will lead to fixed percentage overshoots .	Remember	CO3	CLO10	AEC012.10
42	What is IIR Property?	Infinite impulse response (IIR) is a property applying to many linear time-invariant systems.	Understand	CO3	CLO10	AEC012.10
43	What makes a filter IIR?	IIR filters are digital filters with infinite impulse response. Unlike FIR filters, they have the feedback (a recursive part of a filter) and are known as recursive digital filters	Understand	CO3	CLO7	AEC012.07
44	What is the significance of IIR?	Infinite impulse response (IIR) is a property applying to many linear time-invariant systems. linear time-invariant systems are most electronic and digital filters.	Understand	CO3	CLO8	AEC012.08
45	Why Digital filters solution for analog filters?	digital IIR filters can be based on well-known solutions for analog filters such as the Chebyshev filter, Butterworth filter, and elliptic filter, inheriting the characteristics of those solutions.	Understand	CO3	CLO9	AEC012.09

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
46	What is analog filter	An analog filter that removes all signals below a certain frequency is a high pass filter, because it lets pass everything higher than the cutoff frequency. Digital filters are often embedded in a chip that operates on digital signals, such as an MCU, SoC, processor, or DSP	Understand	CO3	CLO10	AEC012.10
47	Define IIR system.	If the impulse of the system is infinite durarion then the system is called infinite impulse response(IIR) system.	Understand	CO3	CLO11	AEC012.11
48	What is a filter?	Filter is system which removes unwanted frequencies from the input signal	Remember	CO3	CLO11	AEC012.11
49	Define IIR system.	If the impulse of the system is infinite durarion then the system is called infinite impulse response(IIR) system.	Understand	CO3	CLO7	AEC012.07
50	What is warping effect	The effect of the non-linear compression at high frequencies is called warping effect.	Understand	CO3	CLO8	AEC012.08
		UNIT-IV				
1	What is Half-band filter?	A type of FIR filter where the transition region is centered at one quarter of the sampling rate,	Understand	CO4	CLO12	AEC012.12
2	What is Transversal Filter?	It is another name for standard FIR filter implementations, where the input samples traverse their way through the delay elements of a FIR filter.	Remember	CO4	CLO13	AEC012.13
3	What is Parks-McClellan Method?	The Parks-McClellan method (inaccurately called "Remez" by Matlab) is probably the most widely used FIR filter design method.	Remember	CO4	CLO14	AEC012.14
4	What is LPF?	A low-pass filter (LPF) is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency	Understand	CO4	CLO12	AEC012.12
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 w, which inturn requires constant group delay and phase delay.	Remember	CO4	CLO13	AEC012.13
6	What is HPF?	A high-pass filter (HPF) is an electronic filter that passes signals with a frequency higher than a certain cutoff frequency and attenuates signals with frequencies lower than the cutoff frequency.	Remember	CO4	CLO14	AEC012.14
7	What is BPF?	A band-pass filter or BPF, is a device that passes frequencies	Understand	CO4	CLO12	AEC012.12

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		within a certain range and rejects (attenuates) frequencies outside that range.				
8	What is BSF?	a band-stop filter or band-rejection filter is a filter that passes most frequencies unaltered, but attenuates those in a specific range to very low levels.	Remember	CO4	CLO13	AEC012.13
9	What is transfer function?	A transfer function (also known as system function or network function) of a system component gives the device's output for each possible input.	Understand	CO4	CLO14	AEC012.14
10	What is signal flow graph?	A specialized flow graph, a directed graph in which nodes represent system variables, and branches (edges, arcs, or arrows) represent functional connections between pairs of nodes	Understand	CO4	CLO12	AEC012.12
11	What is delay locked loop?	A DLL compares the phase of its last output with the input clock to generate an error signal which is then integrated and fed back as the control to all of the delay elements.	Remember	CO4	CLO13	AEC012.13
12	What is all pass filter?	An all-pass filter is a signal processing filter that passes all frequencies equally in gain, but changes the phase relationship among various frequencies	Understand	CO4	CLO14	AEC012.14
13	What is notch filter?	A filter that attenuates signals within a very narrow band of frequencies. These filtersreject/attenuate signals in a specific frequency band	Understand	CO4	CLO12	AEC012.12
14	What is lattice structure?	The filter structure is feedforward and the net impulse response is finite length. Lattice filters are used in a variety of adaptive filter applications	Understand	CO4	CLO13	AEC012.13
15	What is digital sine-cosine generator?	A Digital Sine-Cosine Generator generates a sine and cosine waves based on FPGA. These signals can be used to generate a PWM signals that can be used in SDR and DSP.	Understand	CO4	CLO14	AEC012.14
16	What is mixed signal device?	It collects analog signals and converts them into digital data to be processed. Once a DSP processes and compresses the digital data, a mixed-signal device decompresses, transmits and displays the digital data.	Understand	CO4	CLO12	AEC012.12

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
17	Why sampling method in FIR filters?	It is suitable for narrow band frequency selective filters only.	Understand	CO4	CLO13	AEC012.13
18	What is Kaiser window?	The Kaiser window is an approximation to the prolate spheroidal window, for which the ratio of the mainlobe energy to the sidelobe energy is maximized. For a Kaiser window of a particular length, the parameter β controls the sidelobe height.	Remember	CO4	CLO14	AEC012.14
19	What is hamming window?	The Hamming window is a taper formed by using a raised cosine with non-zero endpoints, optimized to minimize the nearest side lobe.	Remember	CO4	CLO12	AEC012.12
20	What is FIR filter optimization ?	A very efficient technique for drastically reducing the number of multipliers and adders in implementing linear-phase finite impulse response filters.	Understand	CO4	CLO13	AEC012.13
21	What is Anti-Symmetric FIR filter?	FIR filter with unit impulse response or unit sample response as h(n)= -h(M-1-n), for n=0,1,2...M-1	Understand	CO4	CLO14	AEC012.14
22	What is FIR tap?	A FIR "tap" is simply a coefficient/delay pair. The number of FIR taps, (often designated as "N") is an indication of 1) the amount of memory required to implement the filter, 2) the number of calculations required.	Remember	CO4	CLO12	AEC012.12
23	Define Symmetric FIR filter.	FIR filter with unit impulse response or unit sample response as h(n)= h(M-1-n), for n=0,1,2…M-1	Understand	CO4	CLO13	AEC012.13
24	What is transition band?	The band of frequencies between pass band and stop band edges. The narrower the transition band, the more taps are required to implement the filter. (A "small" transition band results in a "sharp" filter.)	Understand	CO4	CLO14	AEC012.14
25	What is Hilbert transform?	The transform technique in which phase angle of all components of the signal is shifted by ±900 (degrees). The Hilbert transformer is very useful when out of phase component (or imaginary part)	Understand	CO4	CLO12	AEC012.12

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		need to be generated from available real component of the signal.				
26	What is LMS algorithm?	Least mean squares (LMS) algorithms are a class of adaptive filters used to mimic (imitate) a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal	Understand	CO4	CLO13	AEC012.13
27	Define sampling.	The process of converting continuous time analog signal into discrete samples is called sampling .	Understand	CO4	CLO14	AEC012.14
28	What is poly-phase realization?	Poly-phase is a way of doing sampling-rate conversion that leads to very efficient implementations. But more than that, it leads to very general viewpoints that are useful in building filter banks.	Understand	CO4	CLO12	AEC012.12
29	Define Pass band.	A pass band is the range of frequencies or wavelengths that can pass through a filter.	Understand	CO4	CLO13	AEC012.13
30	Define Attenuation.	Attenuation is a general term that refers to any reduction in the strength of a signal. Attenuation occurs with any type of signal, whether digital or analog.	Understand	CO4	CLO14	AEC012.14
31	What is Ripple values ?	The limits of the tolerances in the pass band and stop band are called ripple values.	Understand	CO4	CLO12	AEC012.12
32	What is Stop band ?	A stop band is a band of frequencies, between specified limits, through which a circuit, such as a filter or telephone circuit, does not allow signals to pass, or the attenuation is above the required stop band attenuation level.	Remember	CO4	CLO13	AEC012.13
33	What is Stop band attenuation?	A stop band is a band of frequencies, between specified limits, through which a circuit, such as a filter or telephone circuit, does not allow signals to pass, or the attenuation is above the required stop band attenuation level.	Understand	CO4	CLO14	AEC012.14
34	What is Pass band ripple?	The pass band ripple is the amount of variation in the amplitude, within the designated pass band of the filter,	Remember	CO4	CLO12	AEC012.12
35	What is a Bessel Function.	A mathematical function used to produce the most linear phase response of all IIR filters with no consideration of the frequency magnitude response.	Understand	CO4	CLO13	AEC012.13

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
36	What is Butterworth Function?	A mathematical function used to produce maximally flat filter magnitude responses with no consideration of phase linearity or group delay variations.	Remember	CO4	CLO14	AEC012.14
37	What is a Cascaded filters?	The implementation of a filtering system where multiple individual filters are connected in series.	Understand	CO4	CLO12	AEC012.12
38	Define envelope delay distortion.	Signal distortion that results when the rate of change of phase Shift with frequency over the necessary bandwidth of the signal is not constant.	Remember	CO4	CLO12	AEC012.12
39	What is Quadrature Filter?	a dual-path digital filter operating on complex signal sequence, One filter operates on the in-phase data while the other filter processes the quadrature-phase signal data.	Understand	CO4	CLO13	AEC012.13
40	What is a FIR system?	FIR filters are one of two primary types of digital filters used in Digital Signal Processing. "FIR" means "Finite Impulse Response."	Remember	CO4	CLO14	AEC012.14
41	What is linear phase?	phase response of the filter is a linear function of frequency. The result is that all frequency components of the input signal are shifted in time (usually delayed) by the same constant amount (the slope of the linear function),	Understand	CO4	CLO12	AEC012.12
42	What is group delay?	A more commonly encountered representation of filter phase response is called the group delay, i.e all frequency components of the input signal are shifted in time (usually delayed) by the same constant amount	Understand	CO4	CLO13	AEC012.13
43	What is phase delay?	The phase delay gives the time delay in seconds experienced by each component of the input signal. The phase delay expresses the phase response as a time delay in seconds.	Understand	CO4	CLO14	AEC012.14
44	Phase unwrapping means?	Phase unwrapping ensures that all appropriate multiples of 2π, have been included in phase response. We defined phase response simply as the complex angle of the frequency response , and this is not sufficient for obtaining a phase response which can be converted to true time delay.	Remember	CO4	CLO12	AEC012.12
45	Define limit cycle?	Nonlinearities may cause an IIR filter, which is stable under	Understand	CO4	CLO13	AEC012.13

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		infinite precision, to exhibit an unstable behavior under finite precision arithmetic for specific input signals. This type of instability usually results in an oscillatory periodic output called a limit cycle				
46	What is Gibbs phenomenon?	Abrupt truncation of infinite series is equivalent to multiplying infinite series with rectangular sequence .This type of truncation may result in poor convergence of the series. i.e at the point of discontinuity some oscillation may be observed in resultant output signal .This oscillation or ringing is generated because of side lobes in the frequency response. This oscillatory behavior is called "Gibbs Phenomenon".	Remember	CO4	CLO12	AEC012.12
47	What is Windowing Technique?	Windowing is the quickest method for designing an FIR filter. A windowing function simply truncates the ideal impulse response to obtain a causal FIR approximation that is non- causal and infinitely long. Smoother window functions provide higher out-of band rejection in the filter response. However this smoothness comes at the cost of wider stop band transitions	Remember	CO4	CLO13	AEC012.13
48	What is a equi ripple filter?	An equi-ripple filter is simply a filter with ripples of equal height. The magnitude response of actual digital filters may exhibit ripples These ripples are the manifestation of the Gibbs phenomenon. The fact that equi ripple filters have ripples of equal height should not mean much. It is more important that the design of equiripple filters is such that the height of these ripples can be controlled. This itself is not unique of equi ripple filters.	Understand	CO4	CLO14	AEC012.14
49	What is a window?	Windows are sometimes used in the design of digital filters, in particular to convert an "ideal" impulse response of infinite duration, such as a sinc function, to a finite impulse response (FIR) filter design. That is called the window method.	Remember	CO4	CLO12	AEC012.12

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
50	What is Anti-symmetric FIR filter?	FIR filter with unit impulse response or unit sample response as h(n)= -- h(M-1-n), for n=0,1,2…………….M-1	Understand	CO4	CLO13	AEC012.13
UNIT-V						
1	Define adder overflow limit cycle	With fixed-point arithmetic it is possible for filter calculations to overflow. The term overflow oscillation, sometimes also called adder overflow limit cycle, refers to a high-level oscillation that can exist in an otherwise stable filter due to the nonlinearity associated with the overflow of internal filter calculations	Understand	CO5	CLO15	AEC012.15
2	What is Floating-point quantization error?	With floating-point arithmetic it is necessary to quantize after both multiplications and additions. The addition quantization arises because, prior to addition, the mantissa of the smaller number in the sum is shifted right until the exponent of both numbers is the same. In general, this gives a sum mantissa that is too long and so must be quantized	Remember	CO5	CLO16	AEC012.16
3	Define Limit cycle due to product round-off error	A limit cycle, sometimes referred to as a multiplier roundoff limit cycle, is a low-level oscillation that can exist in an otherwise stable filter as a result of the nonlinearity associated with rounding (or truncating) internal filter calculations. Limit cycles require recursion to exist and do not occur in nonrecursive FIR filters.	Understand	CO5	CLO17	AEC012.17
4	Define scaling in signal processing	Scaling is often used to constrain the dynamic range of the variables to a certain word-length	Remember	CO5	CLO15	AEC012.15
5	What is the principle of scaling?	A process of readjusting certain internal gain parameters in order to constrain internal signals to a range appropriate to the hardware with the constraint that the transfer function from input to output should not be changed	Understand	CO5	CLO16	AEC012.16
6	Define deadband effects ?	A deadband (sometimes called a neutral zone or deadzone) is a band of input values in the domain of a transfer function in signal processing system where the output is zero (the output is 'dead' - no action occurs)	Remember	CO5	CLO17	AEC012.17

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
7	What are the drawbacks of IIR filters	Phase distortion and ringing	Remember	CO5	CLO15	AEC012.15
8	Define Frequency Transforms of Decimated and Expanded Sequence.	The analysis of decimation and expansion is better understood by assessing their respective frequency spectrums using the Fourier transform.	Understand	CO5	CLO16	AEC012.16
9	Define Fixed-Point Quantization Errors.	In fixed-point arithmetic, a multiply doubles the number of significant bits. But word size is fixed, which gives error.	Remember	CO5	CLO17	AEC012.17
10	Define the Over Sampling.	If this data were converted directly into an analogue signal, image frequency bands centred on multiples of the sampling-rate would occur, causing amplifier overload, and distortion in the music signal. To protect against this, a common technique called oversampling	Understand	CO5	CLO15	AEC012.15
11	What is PDSP?	Programmable digital signal processors (PDSPs) are microprocessors that are specialized to perform well in digital signal processing intensive applications. A standard microprocessor can do most Programmable DSP operations.	Remember	CO5	CLO15	AEC012.15
12	What is radar signal processing?	In modern Radar systems digital signal processing (DSP) is used extensively. The front end of the receiver for RADAR is still often analog due the high frequencies involved. With fast ADC convertors- often multiple channel, complex IF signals are digitized.	Understand	CO5	CLO16	AEC012.16
13	What is quantization noise?	Quantization noise is a model of quantization error introduced by quantization in the analog-to-digital conversion (ADC) in telecommunication systems and signal processing. It is a rounding error between the analog input voltage to the ADC and the output digitized value. The noise is non-linear and signal-dependent.	Remember	CO5	CLO15	AEC012.15
14	Define spectrum	According to Fourier analysis, any physical signal can be decomposed into a number of discrete frequencies, or a spectrum of frequencies over a continuous range. The statistical	Understand	CO5	CLO16	AEC012.16

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		average of a certain signal or sort of signal (including noise) as analyzed in terms of its frequency content, is called its spectrum.				
15	What are Finite word-length effects?	There are number of effects of finite word length like overflow error in addition, round off error in multiplication, effects of coefficient quantization, limit cycle, etc.	Remember	CO5	CLO15	AEC012.15
16	What is down sampling?	In digital signal processing, down sampling is associated with the process of resampling in a multi-rate digital signal processing system.	Understand	CO5	CLO16	AEC012.16
17	Define up sampling.	In the domain of digital signal processing, the term interpolation refers to the process of converting a sampled digital signal (such as a sampled audio signal) to that of a higher sampling rate (Up sampling) using various digital filtering techniques (e.g., convolution with a frequency-limited impulse signal).	Remember	CO5	CLO17	AEC012.17
18	List out the applications of MultiMate Digital Signal Processing	Phase shifters, Analog to Digital Converter. Digital to Analog Converter.	Application	CO5	CLO15	AEC012.15
19	Define Round-off noise Power.	Round off noise is that error in the filter output that results from rounding or truncating calculations within the filter. As the name implies, this error looks like low-level noise at the filter output.	Understand	CO5	CLO16	AEC012.16
20	Define coefficient quantization error.	Discretization (quantization) of the filter coefficients has the effect of perturbing the location of the filter poles and zeroes. As a result, the actual filter response differs slightly from the ideal response. This deterministic frequency response error is referred to as coefficient quantization error	Remember	CO5	CLO17	AEC012.17
21	What is rational sampling rate conversion?	In Decimation and Interpolation, sampling rate conversion is achieved by Integer Factor. When sampling rate conversion requires by non integer factor, we need to perform sampling rate conversion by rational factor I/D.	Remember	CO5	CLO15	AEC012.15
22	What is meant by fixed point number?	In fixed point number the position of a binary point is fixed. The bit to the right	Understand	CO5	CLO16	AEC012.16

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		represent the fractional part and those to the left is integer part.				
23	What is product quantization error?	The product quantization errors arise at the output of the multiplier. Multiplication of a b bit data with a b bit coefficient results a product having 2b bits. Since a b bit register is used the multiplier output will be rounded or truncated to b bits which produce the error.	Remember	CO5	CLO17	AEC012.17
24	What are finite word length effects?	Practical digital filters must be implemented with finite precision numbers and arithmetic. As a result, both the filter coefficients and the filter input and output signals are in discrete form. This leads to finite word length effects.	Understand	CO5	CLO17	AEC012.17
25	What is truncation?	Truncation is a process of discarding all bits less significant than LSB that is retained	Remember	CO5	CLO15	AEC012.15
26	What is Rounding?	Rounding a number to b bits is accomplished by choosing a rounded result as the b bit number closest number being unrounded.	Understand	CO5	CLO16	AEC012.16
27	What is floating point arithmetic?	In the past, many DSP applications used fixed point arithmetic due to the high cost (in delay, silicon area, and power consumption) of floating-point arithmetic units. This work extends the consideration of fused floating-point arithmetic to operations that are frequently encountered in DSP.	Remember	CO5	CLO17	AEC012.17
28	What is coefficient inaccuracy?	The Effect of Coefficient Quantization on the Performance of a Digital Filter. As a result, after implementing a filter, we may observe that the frequency response of the filter is quite different from that of the original design. The error in the pole and zero locations depends on several factors	Understand	CO5	CLO15	AEC012.15
29	What is fixed point arithmetic?	Pipelines and additional data paths and registers have also been added to speed and automate arithmetic operations and data transfers. DSP processors fall into two major categories based on the way they represent numerical values.	Remember	CO5	CLO16	AEC012.16

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
30	What is coefficient quantization error?	The Effect of Coefficient Quantization on the performance of a digital filter. As a result, after implementing a filter, the frequency response of the filter is quite different from that of the original design. The error in the pole and zero locations depends on several factors	Understand	CO5	CLO17	AEC012.17
31	What are the quantization methods?	The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as Quantization. Truncation Rounding	Remember	CO5	CLO15	AEC012.15
32	What is product round-off error?	A round off error, also called rounding error, is the difference between the result produced by a given algorithm using exact arithmetic and the result produced by the same algorithm using finite-precision, rounded arithmetic.	Understand	CO5	CLO16	AEC012.16
33	What is quantization step size?	Quantization step size, in mathematics and digital signal processing, is the process of mapping input values from a large set (often a continuous set) to output values in a (countable) smaller set, often with a finite number of elements.	Remember	CO5	CLO17	AEC012.17
34	What is a zero-input oscillation?	In DSP, in the study of dynamical systems with two-dimensional phase space, a limit cycle is a closed trajectory in phase space having the property that at least one other trajectory spirals into it either as time approaches infinity or as time approaches negative infinity.	Understand	CO5	CLO15	AEC012.15
35	What is complex multi-rate system?	A multi-rate DSP system uses multiple sampling rates within the system. Whenever a signal at one rate has to be used by a system that expects a different rate, the rate has to be increased or decreased, and some processing is required to do so.	Remember	CO5	CLO16	AEC012.16
36	Define Interpolation filter.	Since up-sampling causes periodic repetition of the basic spectrum, the unwanted images in the spectra of the up-sampled signal must be removed by using a low pass filter H(z).	Understand	CO5	CLO17	AEC012.17
37	Define decimation.	Decimation is the process of reducing the sampling rate. In	Remember	CO5	CLO15	AEC012.15

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		practice, this usually implies low pass-filtering a signal, then throwing away some of its samples.				
38	What is aliasing step?	In signal processing and related disciplines, aliasing is an effect that causes different signals to become indistinguishable (or aliases of one another) when sampled.	Understand	CO5	CLO16	AEC012.16
39	What is anti-aliasing filter?	An anti-aliasing filter is a filter used before a signal sampler to restrict the bandwidth of a signal to approximately or completely satisfy the sampling theorem over the band of interest.	Remember	CO5	CLO17	AEC012.17
40	What is Delta-sigma quantizer?	The coarsely-quantized output of a delta-sigma modulator is occasionally used directly in signal processing or as a representation for signal storage.	Understand	CO5	CLO15	AEC012.15
41	Define quantization.	A few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as Quantization.	Remember	CO5	CLO16	AEC012.16
42	Define round-off error.	A round off error, also called rounding error, is the difference between the result produced by a given algorithm using exact arithmetic and the result produced.	Understand	CO5	CLO17	AEC012.17
43	What is quantization step size?	Quantization step size, in mathematics and digital signal processing, is the process of mapping input values from a large set to output values in a smaller set, often with a finite number of elements.	Remember	CO5	CLO15	AEC012.15
44	What is a zero-input oscillation?	In the study of dynamical systems with two-dimensional phase space, a limit cycle is a closed trajectory in phase space having the property that at least one other trajectory spirals into it either as time approaches infinity or as time approaches negative infinity.	Understand	CO5	CLO16	AEC012.16
45	What is complex multirate system?	A multi-rate DSP system uses multiple sampling rates within the system. Whenever a signal at one rate has to be used by a system that expects a different rate, the rate has to be increased or decreased.	Remember	CO5	CLO17	AEC012.17
46	What is multi-rate DSP?	Multi rate simply means "multiple sampling rates". A	Understand	CO5	CLO15	AEC012.15

S.No	QUESTION	ANSWER	Blooms Level	CO	CLO	CLO Code
		multi rate DSP system uses multiple sampling rates within the system. Whenever a signal at one rate has to be used by a system that expects a different rate, the rate has to be increased or decreased, and some processing is required to do so.				
47	What is sub-band coding?	In signal processing, SBC is any form of transform coding that breaks a signal into a number of different frequency bands, typically by using a fast Fourier transform, and encodes each one independently.	Remember	CO5	CLO16	AEC012.16
48	What is decimator?	In digital signal processing, down sampling and decimation are terms associated with the process of re-sampling in a multi-rate digital signal processing system. A system component that performs decimation is called a decimator.	Understand	CO5	CLO17	AEC012.17
49	What is interpolator?	In the domain of digital signal processing, the term interpolation refers to the process of converting a sampled digital signal to that of a higher sampling rate using various digital filtering techniques	Remember	CO5	CLO15	AEC012.15
50	What is phase restoration?	Digital filters are a very important part of DSP. In fact, their extraordinary performance is one of the key reasons that DSP has become so popular. Filters have two uses: signal separation and signal restoration. Signal restoration is used when a signal has been distorted in some way	Understand	CO5	CLO16	AEC012.16

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