JEPPIAAR
ENGINEERING COLLEGE

DEPARTMENT OF COMPUTER SCIENCE & ENGINEERING

IT6502

DIGITAL SIGNAL PROCESSING

Question Bank

III YEAR A & B / 2013 REQULATION

BATCH: 2016-2020

Vision of Institution

To build Jeppiaar Engineering College as an Institution of Academic Excellence in Technical education and Management education and to become a World Class University.

Mission of Institution

M1	To excel in teaching and **learning, research and innovation** by promoting the principles of scientific analysis and creative thinking
M2	To participate in the production, **development and dissemination of knowledge** and interact with **national and international communities**
M3	To equip students with **values, ethics and life skills** needed to enrich their lives and enable them to meaningfully contribute to the **progress of society**
M4	To prepare students **for higher studies and lifelong learning**, enrich them with the **practical and entrepreneurial skills** necessary to excel as future professionals and contribute to **Nation's economy**

Program Outcomes (POs)

PO1	**Engineering knowledge**: Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.
PO2	**Problem analysis**: Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.
PO3	**Design/development of solutions**: Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations
PO4	**Conduct investigations of complex problems**: Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.
PO5	**Modern tool usage**: Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modeling to complex engineering activities with an understanding of the limitations.

PO6	**The engineer and society**: Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
PO7	**Environment and sustainability**: Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.
PO8	**Ethics**: Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
PO9	**Individual and team work**: Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.
PO10	**Communication**: Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.
PO11	**Project management and finance**: Demonstrate knowledge and understanding of the engineering and management principles and apply these to one's own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.
PO12	**Life-long learning**: Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

Vision of Department

To emerge as a globally prominent department, developing ethical computer professionals, innovators and entrepreneurs with academic excellence through quality education and research.

Mission of Department

M1	To create **computer professionals** with an ability to identify and **formulate the engineering problems** and also to provide **innovative solutions** through **effective teaching learning process.**
M2	To **strengthen the core-competence** in computer science and engineering and to create an ability to **interact** effectively with industries.
M3	To produce engineers with good professional skills, **ethical values** and life skills for the **betterment of the society.**
M4	To encourage students towards **continuous and higher level learning** on technological advancements and provide a platform for **employment and self-employment.**

Program Educational Objectives (PEOs)

PEO1	**To address the real time complex engineering problems using innovative approach with strong core computing skills.**
PEO2	**To apply core-analytical knowledge and appropriate techniques and provide solutions to real time challenges of national and global society**
PEO3	**Apply ethical knowledge for professional excellence and leadership for the betterment of the society.**
PEO4	**Develop life-long learning skills needed for better employment and entrepreneurship**

Programme Specific Outcome (PSOs)

PSO1 – An ability to understand the core concepts of computer science and engineering and to enrich problem solving skills to analyze, design and implement software and hardware based systems of varying complexity.

PSO2 - To interpret real-time problems with analytical skills and to arrive at cost effective and optimal solution using advanced tools and techniques.

PSO3 - An understanding of social awareness and professional ethics with practical proficiency in the broad area of programming concepts by lifelong learning to inculcate employment and entrepreneurship skills.

BLOOM TAXANOMY LEVELS

BTL1: Remembering
BTL2: Understanding
BTL3Applying.
BTL4: Analyzing
BTL5:Evaluating
BTL6:Creating

SYLLABUS

IT6502 DIGITAL SIGNAL PROCESSING L T P C

3 1 0 4

OBJECTIVES:

- To introduce discrete Fourier transform and its applications.
- To teach the design of infinite and finite impulse response filters for filtering undesired signals.
- To introduce signal processing concepts in systems having more than one sampling frequency.

UNIT I SIGNALS AND SYSTEMS 9

Basic elements of DSP – concepts of frequency in Analog and Digital Signals – sampling theorem – Discrete – time signals, systems – Analysis of discrete time LTI systems – Z transform – Convolution – Correlation.

UNIT II FREQUENCY TRANSFORMATIONS 9

Introduction to DFT – Properties of DFT – Circular Convolution - Filtering methods based on DFT – FFT Algorithms - Decimation – in – time Algorithms, Decimation – in – frequency Algorithms – Use of FFT in Linear Filtering – DCT – Use and Application of DCT.

UNIT III IIR FILTER DESIGN 9

Structures of IIR – Analog filter design – Discrete time IIR filter from analog filter – IIR filter design by Impulse Invariance, Bilinear transformation, Approximation of derivatives – (LPF, HPF, BPF, BRF) filter design using frequency translation.

UNIT IV FIR FILTER DESIGN 9

Structures of FIR – Linear phase FIR filter – Fourier Series - Filter design using windowing techniques (Rectangular Window, Hamming Window, Hanning Window), Frequency sampling techniques

UNIT V FINITE WORD LENGTH EFFECTS IN DIGITAL FILTERS 9

Binary fixed point and floating point number representations – Comparison - Quantization noise – truncation and rounding – quantization noise power- input quantization error- coefficient quantization error – limit cycle oscillations-dead band- Overflow error-signal scaling.

TOTAL (L:45+T:15): 60 PERIODS

OUTCOMES:

Upon completion of the course, students will be able to:

- Perform frequency transforms for the signals.
- Design IIR and FIR filters.
- Finite word length effects in digital filters

TEXT BOOK:

1. John G. ProaKis and Dimitris G.ManolaKis, “Digital Signal Processing – Principles, Algorithms & Applications”, Fourth Edition, Pearson Education, Prentice Hall, 2007.

REFERENCES:

1. Emmanuel C.Ifeachor, and Barrie.W.Jervis, “Digital Signal Processing”, Second Edition, Pearson Education, Prentice Hall, 2002.
2. Sanjit K. Mitra, “Digital Signal Processing – A Computer Based Approach”, Third Edition, Tata Mc Graw Hill, 2007.
3. A.V.Oppenheim, R.W. Schafer and J.R. BucK, Discrete-Time Signal Processing, 8th Indian Reprint, Pearson, 2004.
4. Andreas Antoniou, “Digital Signal Processing”, Tata McGraw Hill, 2006.

Course Outcomes (COs)

C312.1	**Understand** the various signals and systems.
C312.2	**Build** frequency transformations for the signals and **Compare** Discrete Fourier Transform and Fast Fourier Transform.
C312.3	**Design** of Infinite Impulse Response filters for given specifications.
C312.4	**Design** of Finite Impulse Response filters for given specifications
C312.5	**Determine** the effects of Finite Word length Effects in Digital Filters.

INDEX PAGE

UNIT	REFERENCE BOOK	PAGE NUMBER
I	John G. ProaKis and Dimitris G.ManolaKis, “Digital Signal Processing – Principles, Algorithms & Applications”, Fourth Edition, Pearson Education, Prentice Hall, 2007	9-89
II	John G. ProaKis and Dimitris G.ManolaKis, “Digital Signal Processing – Principles, Algorithms & Applications”, Fourth Edition, Pearson Education, Prentice Hall, 2007	90-132
III	John G. ProaKis and Dimitris G.ManolaKis, “Digital Signal Processing – Principles, Algorithms & Applications”, Fourth Edition, Pearson Education, Prentice Hall, 2007	133-197
IV	John G. ProaKis and Dimitris G.ManolaKis, “Digital Signal Processing – Principles, Algorithms & Applications”, Fourth Edition, Pearson Education, Prentice Hall, 2007	198-282
V	John G. ProaKis and Dimitris G.ManolaKis, “Digital Signal Processing – Principles, Algorithms & Applications”, Fourth Edition, Pearson Education, Prentice Hall, 2007	283-401

UNIT I PART A

Q. No.	Questions	CO	Bloom's Level
1	**Find the Z transform of {1,0,2,0,3}** **May/ June 2007** 1+0z-1+2z-2+3Z-4	**C312.1**	BTL1
2	**Check whether the system y(n)=ex(n) is linear.** **May/ June 2007** The system is nonlinear.	**C312.1**	BTL 2
3	**What are the advantages of DSP?** **Nov/Dec 2009** 1.More accuracy 2.It is easier to perform mathematical operation 3. Digital signals can be easily stored on magnetic disk without any loss of information.	**C312.1**	BTL 1
4	**Define STEP signal.** **Nov/Dec 2009** Also called as delta function Represented by S(n) S(n)=1 for n is equal to 0 =0 for n is greater than 0	**C312.1**	BTL 1
5	**Define impulse signal.** **Nov/Dec 2009** Also called as delta function Represented by S(n) S(n)=1 for n is equal to 0 =0 for n is not equal to 0	**C312.1**	BTL 1
6	**Find the period of x(n) = cos [8πn/7 +2].** ω=8π/7 2πf = 8π/7 f= 4/7 ; here K= 4 & N =7	**C312.1**	BTL 1
7	**What is meant by causal & non causal system?** A system is said be causal if it's output at anytime depends upon present and past input only. A system is said be non causal if it's output at anytime depends upon present and future input only**.**	**C312.1**	BTL 1
8	**What is the condition for the BIBO stable?** The condition for the BIBO stable is given by $\int^{\infty} \lvert h(t) \rvert dt < ^{a}$	**C312.1**	BTL 1
9	**Distinguish between linear Time Invarient and non linear system. Nov/Dec 2010** $a_1 y_1(t) + a_2 y_2(t) = f[a_1x_1(t) + a_2x_2(t)]$ If the above equation satisfies then the system is said to be Linear system. If the above equation does not satisfies then the system is said to be non	**C312.1**	BTL4

	Linear system**.**		
10	**What are energy and power signals? May /June 2013,Nov/Dec 2012** The energy signal is one in which has finite energy and zero average power The power signal is one in which has finite average power and infinite energy . $E = \underset{T\to\infty}{Lt} \int_{-T}^{T} \lvert x(t) \rvert^2$ dt joules . $P = \underset{T\to\infty}{Lt} \; 1/2T \int^{T} \lvert x(t) \rvert^2$ dt joules .**]**	**C312.1**	BTL 1
11	**What is correlation? What are its types? May /June 2013** Measuring similarities between two signals .Two types are Auto Correlation and Cross Correlation**.**	**C312.1**	BTL 1
12	**Compare linear convolution and circular convolution. Nov/Dec 2012 Nov/Dec 2010** y(n)=(N1+N2-1)samples -input sequence may have different length-Zero padding is not required------Linear convolution, y(n)=max(N1+N2)- input sequence should have same length –If the length of the sequence are not equal Zeroes are appended at the end of the sequence.-----Circular convolution	**C312.1**	BTL 5
13	**What is sampling theorem? Nov/Dec 2012, APRIL/MAY2015** Fs>=2Fm Fs= Sampling frequency Fm- maximum analog frequency**.**	**C312.1**	BTL 1
14	**What do you understand by the term signal processing? MAY/JUNE 2014** Processing of signals by systems is called as signal processing	**C312.1**	BTL 1
15	**What is time invarient system? MAY/JUNE 2014.MAY/JUNE 2016** If the input output characteristics of the systems do not change with time ,then the system is referred as time in variant system.	**C312.1**	BTL 1
16	**What is linear and nonlinear systems?** The system is linear if and only if T[a1x1(n)+ a2x2(n)]= a1y1(n)+ a2y2(n)] Where x1(n),x2(n) are arbitrary input signals y1(n), y2(n) are arbitrary output signals a1,a2 are constants.	**C312.1**	BTL 1
17	**What is static and dynamic systems**. A system is static if its output at any instant n depends only on present input but not on past or future input.	**C312.1**	BTL 1

18	**Define Region of Convergence.(ROC) <u>April/May 2018</u>** Since Z transform is an infinite power series it exists only for those values of Z for which X(Z)=attains a finite value.	**C312.1**	BTL 1
19	**What are the properties of Z transform**. 1.linearity property 2.scaling property3.Time shifting property4.Time reversal property5.Convolution of 2 sequences.6.Differentiation in Z domain	**C312.1**	BTL 1
20	**Write the cases in long division method.** Case1.When ROC exterior to the circle, the system is expected to be a causal system. Case2.When ROC interior to the circle, the system is expected to be a anticausal system..	**C312.1**	BTL 1
21	**what are the types of convolution.** 1.circular convolution2.linear convolution	**C312.1**	BTL 1
22	**What are the types of correlation?** Autocorrelation- measuring similarities between same signals Cross correlation-measuring similarities between different signals	**C312.1**	BTL 1
23	**Define DSP.** Processing of signals by digital systems	**C312.1**	BTL 1
24	**Find the energy of (1/4)n u(n) <u>April/May 2017</u>** Refer Notes	**C312.1**	BTL 1
25	**What are the types of signals.** 1.one dimensional signals2. multi dimensional signals3. multi channel signals	**C312.1**	BTL 1
26	**What is Nyquest sampling Rate?** Fs>=2Fm Sampling frequency should be greater than two times maximum frequency.	**C312.1**	BTL 1
27	**What is continues time signals AND Discrete time signals.** Amplitude can be defined for all values of t. Amplitude can be defined for particular integer values of t.**<u>MAY/JUNE 2016</u>**	**C312.1**	BTL 2
29	**Given x(z)=Z2 +2Z +1-2Z-2. Find the equivalent time domain signal x(n) <u>Nov/Dec2017</u>** **Refer notes**	**C312.1**	BTL 3
30	**What are the applications of DSP?** Image processing, speech processing, biomedical, Radar system, Digital audio, video processing	**C312.1**	BTL 1
31	**What is continuous and discrete time signal**? **Continuous time signal** A signal x(t) is said to be continuous if it is defined for all time t. Continuous time signal arise naturally when a physical waveform such as	**C312.1**	BTL 1

	acoustics wave or light wave is converted into a electrical signal. **Discrete time signal** A discrete time signal is defined only at discrete instants of time. The independent variable has discrete values only, which are uniformly spaced. A discrete time signal is often derived from the continuous time signal by sampling it at a uniform rate		
32	**State distributive law** The distributive law can be expressed as x(n)*[h1(n)+h2(n)]=x(n)*h1(n)+x(n)*h2(n)	**C312.1**	BTL 1
33	**Define 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.	**C312.1**	BTL 1
34	**Define discrete** time system. A discrete or an algorithm that performs some prescribed operation on a discrete time signal is called discrete time system.	**C312.1**	BTL 1
35	**What are the elementary discrete time signals?** • Unit sample sequence (unit impulse) δ (n)= {1 n=0 0 Otherwise • Unit step signal U (n) ={ 1 n>=0 0 Otherwise • Unit ramp signal Ur(n)={n for n>=0 0 Otherwise • Exponential signal x (n)=an where a is real x(n)-Real signal	**C312.1**	BTL 1
36	**Define symmetric and antisymmetric signal**. A real value signal x (n) is called symmetric (even) if x (-n) =x (n). On the other hand the signal is called antisymmetric (odd) if x (-n) =x (n	**C312.1**	BTL 1
37	**Define dynamic and static system.** A discrete time system is called static or memory less if its output at any instant n depends almost on the input sample at the same time but not on past and future samples of the input. e.g. y(n) =a x (n) In anyother case the system is said to be dynamic and to have memory. e.g. (n) =x (n)+3 x(n-1)	**C312.1**	BTL 1
38	**Define linear and non-linear systems** Linear system is one which satisfies superposition principle. Superposition principle: The response of a system to a weighted sum of signals be equal to the corresponding weighted sum of responses of system to each of individual	**C312.1**	BTL 1

	input signal. i.e., T [a1x1(n)+a2x2(n)]=a1T[x1(n)]+a2 T[x2(n)] e.g. y(n)=n x(n) A system which does not satisfy superposition principle is known as non-linear system. e.g.(n)=x2(n)		
39	**What are the steps involved in calculating convolution sum**? The steps involved in calculating sum are • Folding • Shifting • Multiplication • Summation	**C312.1**	BTL 1
40	**state associative law** The associative law can be expressed as [x(n)*h1(n)]*h2(n)=x(n)[h1(n)*h2(n)] Where x(n)-input h1(n)-impulse response. 19.State commutative law The commutative law can be expressed as x(n)*h(n)=h(n)*x(n)	**C312.1**	BTL 1
41	**what are the properties of convolution sum** The properties of convolution sum are • Commutative property. • Associative law. • Distributive law.	**C312.1**	BTL 1
42	**State distributive law** The distributive law can be expressed as x(n)*[h1(n)+h2(n)]=x(n)*h1(n)+x(n)*h2(n)	**C312.1**	BTL 1
43	**State properties of ROC.** • The ROC does not contain any poles. • When x(n) is of finite duration then ROC is entire Z-plane except Z=0 or Z=∞. • If X(Z) is causal,then ROC includes Z=∞. • If X(Z) is anticasual,then ROC includes Z=0.	**C312.1**	BTL 1
44	**How to obtain the output sequence of linear convolution through circular convolution?** Consider two finite duration sequences x(n) and h(n) of duration L samples and M samples. The linear convolution of these two sequences produces an output sequence of duration L+M-1 samples, whereas , the circular convolution of x(n) and h(n) give N samples where N=max(L,M).In order to obtain the number of samples in circular convolution equal to L+M-1, both x(n) and h(n) must be appended with appropriate number of zero valued samples. In other words by increasing the length of the sequences x(n) and h(n) to L+M-1 points and then circularly convolving the resulting sequences	**C312.1**	BTL 1

	we obtain the same result as that of linear convolution.		
45	**What is zero padding?What are its uses?** Let the sequence x(n) has a length L. If we want to find the N-point DFT(N>L) of the sequence x(n), we have to add (N-L) zeros to the sequence x(n). This is known as zero padding. The uses of zero padding are 1)We can get better display of the frequency spectrum. 2)With zero padding the DFT can be used in linear filtering.	**C312.1**	BTL 1
46	**Find the convolution of X(n)=1,2,3,1,2,1 and h(n)=1,2,1, <u>April/May 2018</u>** Refer Notes	**C312.1**	BTL 2
47	**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 cicular 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.	**C312.1**	BTL 1
48	**.What is overlap-save method?** In this method the data sequence is divided into N point sections xi(n).Each section contains the last M-1 data points of the previous section followed by L new data points to form a data sequence of length N=L+M-1.In circular convolution of xi(n) with h(n) the first M-1 points will not agree with the linear convolution of xi(n) and h(n) because of aliasing, the remaining points will agree with linear convolution. Hence we discard the first (M-1) points of filtered section xi(n) N h(n). This process is repeated for all sections and the filtered sections are abutted together.	**C312.1**	BTL 1
49	A signal x(t) =sin (5 **πt) is sampled and what is the minimum sampling frequency is needed to reconstruct the signal without aliasing<u>. Nov/Dec 2018.</u>** Fs>2fa .fa=2.5 therefore sampling frequency should be greater than or equal to 5 kz.	**C312.1**	BTL 1
50	Find the system transfer function of given difference equation using Z transform y(n)-0.5y(n-1)=x(n). **<u>Nov/Dec 2018</u>.** Y(Z)=X(Z). H(Z) Therefore H(Z)= Y(Z)/X(Z).	**C312.1**	BTL 1

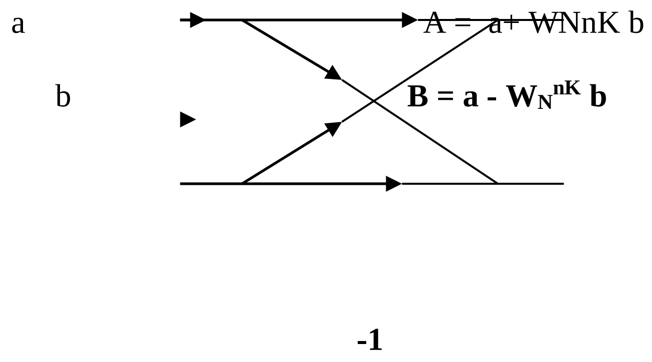
PART-B

Q. No.	Questions	CO	Bloom's Level
1.	**Discover the circular convolution and correlation for $x(n)=\{0,1,-2,3,-4\}$ and $h(n)=\{0.5,1,2,1,0.5\}$ April/May2008, APRIL/MAY2015 Nov/Dec2017.Nov/Dec 2018** Ans. Refer page no 165 DSP by proakis	**C312.1**	BTL 4
2.	**Determine the impulse response of the difference equation** **$Y(n)+3y(n-1)+2y(n-2)=2x(n)-x(n-1)$ April /May 2008** Ans.Refer page no 18 DSP by proakis	**C312.1**	BTL 5
3.	**Find the response of the system for the input signal using linear convolution (8) April/May 2017 .Nov/Dec 2018** **May/June2007** **$X(n)=\{1,2,2,3\}$ and $h(n)=\{1,0,3,2\}$** Ans. Refer page no 164 DSP by proakis (ii) Find the inverse Z transform of **April/May 2017** $1/(1-1/2Z-1)(1-1/4 Z-1)$ Ans. Refer page no 461 DSP by proakis	**C312.1**	BTL 4
4.	**Deduct whether the following systems are linear time invarient April/May 2017 May/June2007** **(**i) $y(n)=A+Bx(n)$ Refer page no 31 DSP by proakis (ii) $Y(n)=ex(n)$ Refer page no 29 DSP by proakis (iii)$Y(n)=A.X(n)+B (x(n-1)$ Refer page no 30 DSP by proakis	**C312.1**	BTL 5
5.	**Test the stability and causality of the following system** **i. $y(n)=\cos x(n)$**)Refer page no 41 DSP by proakis (ii)$y(n)=x(-n-2)$(8) Refer page no 41 DSP by proakis	**C312.1**	BTL 6
6.	**Find the one sided z-transform of discrete sequences generated by mathematically sampling of the following continuous time function Nov/Dec2009** (i)$x(t)=\sin wt$ Refer page no 455 DSP by proakis (8) (ii)$x(t)=\cos wt$ Refer page no 456 DSP by proakis(8)	**C312.1**	BTL4
7	**Find whether the following system are linear Time invarient** **MAY/JUNE 2014, MAY/JUNE 2016** $y(n)=e-x(n)$	**C312.1**	BTL4

8	**Find the Z transform of the following discrete time signals and find ROC x(n)=u(n-2) x(n)=[-1/5]nu(n)+5[1/2]-nu(-n-1) MAY/JUNE 2014, MAY/JUNE 2016** Refer notes	**C312.1**	BTL 1
9	**Explain the process of analog to digital conversion of signal in terms of sampling quantization and coding. APRIL/MAY2015 OR (Relate Nyquest rate criteria and aliasing effect with sampling process. Discuss how aliasing error can be avoided. Nov/Dec 2018.** Refer Notes	**C312.1**	BTL 5
10	**A Discrete time system is represented by the following difference equations y(n)=3y2(n-1)-nx(n)+4x(n-1)-2x(n+1) for n>0 .Determine the system is memoryless , causal, linear shift variant. Justify your answers. Nov/Dec2017** Refer Notes	**C312.1**	BTL 5
11	**A causal system is represented by the following differential equations Y(n)+1/4 Y(n-1)=X(n)+1/2 X(n-1). Find the system function H(Z) and its coreponding region of convergence(ROC) Nov/Dec2017** Refer Notes	**C312.1**	BTL 4
12	**Find the unit sample respose h(n) of the system for the given equation Y(n)+1/4 Y(n-1)=X(n)+1/2 X(n-1) Nov/Dec2017** Refer Notes	**C312.1**	BTL 1
13	**Determine the inverse Ztransform of X(Z)=1/1-1.5 z-1 +0.5 Z-2) if ROC Z>1, ROC Z<0.5 and ROC 0.5<Z<1 Apr/May 2017** Refer Notes	**C312.1**	BTL 5
14	**Find the Z transform and ROC of (i) X(n)=s(n)** **(ii) X(n)=[3(3)n-4(2)n]u(n)** **OR (Determine the region of convergence of the following signal using Ztransform:x(n)=u(-n), x(n)=u(l-n), x(n)=2n U(n). Nov/Dec 2018.)** **Check whether the system y(n)=nX2 (n) is static or dynamic linear or nonlinear, time variant or time invariant ,causal or Non causal April/May 2018** Refer Notes	**C312.1**	BTL 5
15	**Determine the response of the system described by the difference equation y(n)=0.7 y(n-1)-0.12 y(n-2)+x(n-1)+x(n-2) to the input x(n)=n u(n) April/May 2018** Refer Notes	**C312.1**	BTL 5

UNIT II PART A

Q. No.	Questions	CO	Bloom’ Level
1	**Find the DTFT of a sequence x(n) = an u(n). Nov/Dec 2006, MAY/JUNE 2016,** Solution: $x(n) = a^n u(n)$ $X(e^{j\omega}) = \sum_{n=-\infty}^{\infty} x(n) e^{-j\omega n}$ $X(e^{j\omega}) = \sum_{n=-0}^{\infty} a^n e^{-j\omega n}$ $X(e^{j\omega}) = \sum_{n=-0}^{\infty} (ae^{-j\omega})^n$ $X(e^{j\omega}) = \frac{1}{1-a^{-e^{j\omega}}}$	**C312.2**	BTL 1
2	**What is FFT? Nov/Dec 2006** The Fast Fourier Transform is a method or algorithm for computing the DFT with reduced number of calculations. The computational efficiency can be achieved if we adopt a divider and conquer approach. This approach is based on decomposition of an N-point DFT in to successively smaller DFT’s. This approach leads to a family of an efficient computational algorithm is Known as FFT algorithm	**C312.2**	BTL 1
3	**The first five DFT coefficients of a sequence x(n) are X(0) = 20, X(1) 5+j2,X(2) = 0,X(3) = 0.2+j0.4 , X(4) = 0 . Discover the remaining DFT coefficients. May/June 2007 April/May 2017** X (K) = [20, 5+j2, 0, 0.2+j 0.4, 0, X (5), X (6), X (7)] X (5) = 0.2 – j0.4 X (6) = 0 X (7) = 5-j2	**C312.2**	BTL 4
4	**What are the advantages of FFT algorithm over direct computation of DFT? Nov/Dec2017 May/June 2007** Reduces the computation time required by DFT. Complex multiplication required for direct computation is N2 and for FFT calculation is N/2 log 2 N. Speed calculation.	**C312.2**	BTL 1
5	**State and prove Parseval’s Theorem. Nov/Dec 2007** Parseval’s theorem states that If x(n) ↔ X(K) and y(n) ↔ Y(K) , Then	**C312.2**	BTL 2

<table>
<tr><td></td><td>$$\sum_{n=0}^{N-1} x(n) y^*(n) = 1/N \sum_{K=0}^{N-1} X(K) Y^*(K)$$
When y(n) = x(n), the above equation becomes
$$\boxed{\sum^{N-1} |x(n)|^2 = 1/N \sum^{N-1} |X(K)|^2}$$</td><td></td><td></td></tr>
<tr><td>6</td><td>What do you mean by the term “bit reversal” as applied to FFT? <u>Nov/Dec 2007, Apr/May 2011</u>
Re-ordering of input sequence is required in decimation – in –time. When represented in binary notation sequence index appears as reversed bit order of row number.</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>7</td><td>Draw the basic butterfly diagram of radix -2 FFT. April/May <u>2008.,May/June 2013</u>.

 </td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>8</td><td>Distinguish between DIT and DIF –FFT algorithm. <u>Nov/Dec 2008</u>
<table>
<tr><td>S.No</td><td>DIT –FFT Algorithm</td><td>DIF –FFT Algorithm</td></tr>
<tr><td>1.</td><td>The input is in bit reversed order; the output will be normal order.</td><td>The input is in normal ord output will be bit reversed ord</td></tr>
<tr><td>2.</td><td>Each stage of computation the phase factor are multiplied before add subtract operation.</td><td>Each stage of computation the factor are multiplied afte subtract operation.</td></tr>
</table></td><td>C312.2</td><td>BTL 4</td></tr>
<tr><td>9</td><td>If H(K) is the N-point DFT of a sequence h(n) , Prove that H(K) and H(N-K) are complex conjugates. Nov/Dec 2008
This property states that, if h(n) is real , then H(N-K) = H*(K) = H(-K)</td><td>C312.2</td><td>BTL 5</td></tr>
</table>

<table>
<tr><td></td><td>Proof:
By the definition of DFT;
$X(K) = \sum_{n=0}^{N-1} x(n)\, e\, (-j2\pi nK)/N$
Replace ‘K’ by ‘N-K’
$X(N-K) = \sum^{N-1} x(n)\, e\, (-j2\pi n(N-K))/N$
$\boxed{X(N-K) = X^*(K)}$</td><td></td><td></td></tr>
<tr><td>10</td><td>Define DFT pair. <u>May/June 2013</u>
The DFT is defined as
$X(K) = \sum_{n=0}^{N-1} x(n)\, e\, (-j2\pi nK)/N \;;\; K = 0 \text{ to } N-1$
The Inverse Discrete Fourier Transform (IDFT) is defined as
$x(n) = \sum_{K=0}^{N-1} X(K)\, e\, (j2\pi nK)/N \;;\; n = 0 \text{ to } N-1$</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>11</td><td>Distinguish between linear & circular convolution.<table>
<tr><td>Linear convolution</td><td>circular convolution</td></tr>
<tr><td>The length of the input sequence can be different.</td><td>The length of the input sequence should same.</td></tr>
<tr><td>Zero Padding is not required.</td><td>Zero padding is required if the length of sequence is different.</td></tr>
</table></td><td>C312.2</td><td>BTL 4</td></tr>
<tr><td>12</td><td>Why Zero padding is needed? <u>Nov/Dec 2011</u>
Appending zeros to the sequence in order to increase the size or length of the sequence is called zero padding. In circular convolution, when the two input sequence are of different size , then they are converted to equal size by zero padding.</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>13</td><td>Write the shifting property of DFT.
Time shifting property states that DFT $\{x(n-n0)\} = X(K)\, e\, (-j2\pi n0K)/N$</td><td>C312.2</td><td>BTL 1</td></tr>
</table>

<table>
<tr><td>14</td><td>Why do we go for FFT?
The FFT is needed to compute DFT with reduced number of calculations. The DFT is required for spectrum analysis on the spinals using digital computers.</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>15</td><td>What do you mean by radix-2 FFT?
The radix -2 FFT is an efficient algorithm for computing N- point DFT of an N-point sequence .In radix-2 FFT the n-point is decimated into 2-point sequence and the 2-point DFT for each decimated sequence is computed. From the results of 2-point DFT’s, the 4-point DFT’s are computed. From the results of 4 –point DFT’s ,the 8-point DFT’s are computed and so on until we get N - point DFT.</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>16</td><td>Is DFT of a finite duration sequence is periodic? If so state the theorem
Yes .periodic. April/May 2018
Theorem : periodicity property
If x(n)—X(Z)
Then x(n+K)---X(Z+K)</td><td>C312.2</td><td>BTL 4</td></tr>
<tr><td>17</td><td>How many multiplications & addition are involved in radix-2 FFT? <u>(May/June 2012)(Nov/Dec 2010)</u>
For performing radix-2 FFT, the value of N should be such that, N= 2m. The total numbers of complex additions are N log 2 N and the total number of complex multiplication are (N/2) log 2 N.</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>18</td><td>What is Twiddle factor? <u>Nov/Dec 2012,Nov/Dec 2011</u>
Twiddle factor is defined as WN = e –j2π/N. It is also called as weight factor.</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>19</td><td>What is main advantage of FFT ? <u>Nov/Dec 2012,May/June 2012.</u>
FFT reduces the computation time required to compute Discrete Fourier Transform</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>20</td><td>Distinguish between DFT and DTFT. <u>Nov/Dec 2011 & May /June 2012</u>
<table>
<tr><th>S.NO</th><th>DFT</th><th>DTFT</th></tr>
<tr><td>1.</td><td>Obtained by performing sampling operation in</td><td>Sampling is performed only in time domain.</td></tr>
</table></td><td>C312.2</td><td>BTL 4</td></tr>
</table>

<table>
<tr><td></td><td><table>
<tr><td></td><td>both time and frequency domain.</td><td></td></tr>
<tr><td>2.</td><td>Discrete frequency spectrum</td><td>Continuous function of ω</td></tr>
<tr><td>3.</td><td>DFT is denoted by X(K) and is given by
$X(K)=\sum_{n=0}^{N-1} x(n)e^{\frac{-j2\pi nk}{N}} \quad where K = 0 to N-1$</td><td>DTFT is denoted by
$X(\omega)=\sum_{n=-\infty}^{\infty} x(n)e^{-j\omega n}$</td></tr>
<tr><td>4.</td><td>DFT can be applied only to finite length sequence</td><td>DTFT are applicable to any arbitrary sequence.</td></tr>
</table></td><td></td><td></td></tr>
<tr><td>21</td><td>State the advantages of FFT over DFT’s. <u>(Apr/May 2011</u>), May/June 2012. State the Need for using FFT algorithms for computing DFT. <u>NOV/DEC 2017</u>. Reduces the computation time required by DFT .Complex multiplication required for direct computation is N2 and for FFT calculation is N/2 log 2 N .Speed calculation.</td><td>C312.2</td><td>BTL 2</td></tr>
<tr><td>22</td><td>List any two properties of DFT . <u>MAY/JUNE 2014</u>
circular convolution x1(n)*x2(n)=X1(K)X2(K)
Linear property If DFT of x1(n)=X1(K) and DFT of x1(n)=X1(K) then Ax1(n)+bx2(n)=aX1(K)+bX2(K)</td><td>C312.2</td><td>BTL 2</td></tr>
<tr><td>23</td><td>What is meant by radix 2 FFT algorithm? <u>MAY/JUNE 2014</u>, APRIL/MAY2015. N=rm Here r=2 and m=3 the given 8 point sequence is decimated into 2 point sequence. .For each 2 point sequence two point DFT are calculated.</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>24</td><td>Write the methods to perform Linear convolution?
1.Graphical method 2.DFT and IDFT method3.matrix method</td><td>C312.2</td><td>BTL1</td></tr>
<tr><td>25</td><td>Write the methods to perform circular convolution?
1.Graphical method2.DFT and IDFT method3.matrix method</td><td>C312.2</td><td>BTL 1</td></tr>
<tr><td>26</td><td>What is the relationship between Z transform and DFT?
Z Transform is defined by
∞</td><td>C312.2</td><td>BTL 1</td></tr>
</table>

	$X(z)=\sum x(n)z-n$ $n=-\infty$ DFT is defined by $N-1$ $X(K)=\sum x(n)e-j2\pi Kn/N$ where $K=0,1,\ldots N-1$ $n=0$		
27	**State sampling theorem? (Nov 2006 & May/Jun 2009)** Sampling is the process to convert analog time domain continuous signal into discrete time domain signal. But it is the process of converting only time domain not in amplitude domain. Nyquist criteria: We sample the signal based on the following condition i.e., $fs \geq 2fm$ Where fx = Sampling frequency Fm = maximum signal frequency If these above conditions are not satisfied we will meet the following demerits after the sampling process. Guard band2. Aliasing Effect	**C312.2**	BTL 1
28	**What are the applications of FFT algorithms? (May/Jun 2009(R2005)** The applications of FFT algorithm include Linear Filtering (ii) Correlation (iii) Spectrum Analysis	**C312.2**	BTL 1
29	**How many multiplications and additions are required to compute N-point DFT using radix-2 FFT? Assume N=512 April/May 2017** In computing N-point DFT by this method the number of stages of computation will be m-times. The number ‘r’ is called the radix of the FFT algorithms. In radix-2-FFT, the total number of complex additions are reduced to $N \log_2 N$ and total number of complex multiplications are reduced to $(N/2\log_2 N)$	**C312.2**	BTL 2
30	**What is meant by aliasing? How can it be avoided? April/May 2017(Nov 2003)** If we operate the sampler at $fx < fm$, the frequency components of the frequency spectrum will overlap with each other i.e., the lower frequency of the second frequency component will overlap with higher frequency of the	**C312.2**	BTL 1

	first frequency component. This overlapping effect is called as Aliasing effect. For avoiding overlapping of high and low frequency components, we have to use low-pass filter to cut the unwanted high frequency components.		
31	**Why FFT is needed? <u>April/May 2018</u>** The direct evaluation DFT requires N2 complex multiplications and N2 –N complex additions. Thus for large values of N direct evaluation of the DFT is difficult. By using FFT algorithm the number of complex computations can be reduced. So we use FFT.	**C312.2**	BTL 1
32	**How many multiplications are required to compute N point DFT using redix-2 FFT?** The number of multiplications required to compute N point DFT using radix-2 FFT are N log2 N	**C312.2**	BTL 1
33	**How many additions are required to compute N point DFT using redix-2 FFT?** The number of additions required to compute N point DFT using radix-2 FFT are N/2 log2 N respectively,.	**C312.2**	BTL 1
34	**What is DIT algorithm?** Decimation-In-Time algorithm is used to calculate the DFT of a N point sequence. The idea is to break the N point sequence into two sequences, the DFTs of which can be combined to give the DFt of the original N point sequence.This algorithm is called DIT because the sequence x(n) is often splitted into smaller sub- sequences.	**C312.2**	BTL 1
35	**What DIF algorithm?** It is a popular form of the FFT algorithm. In this the output sequence X(k) is divided into smaller and smaller sub-sequences , that is why the name Decimation In Frequency.	**C312.2**	BTL 1
36	**What are the applications of FFT algorithm?** The applications of FFT algorithm includes 1) Linear filtering 2) Correlation 3) Spectrum analysis	**C312.2**	BTL 1
37	**Why the computations in FFT algorithm is said to be in place?** Once the butterfly operation is performed on a pair of complex numbers (a,b) to produce (A,B), there is no need to save the input pair. We can store the result (A,B) in the same locations as (a,b). Since the same storage	**C312.2**	BTL 1

	locations are used throughout the computation we say that the computations are done in place.		
38	.**Distinguish between linear convolution and circular convolution of two sequences.** Linear convolution If x(n) is a sequence of L number of samples and h(n) with M number of samples, after convolution y(n) will have N=L+M-1 samples. It can be used to find the response of a linear filter. Zero padding is not necessary to find the response of a linear filter Circular convolution If x(n) is a sequence of L number of samples and h(n) with M samples, after convolution y(n) will have N=max(L,M) samples. It cannot be used to find the response of a filter.	**C312.2**	BTL 4
39	.**What are differences between overlap-save and overlap-add methods**. Overlap-save method In this method the size of the input data block is N=L+M-1 Each data block consists of the last M-1 data points of the previous data block followed by L new data points In each output block M-1 points are corrupted due to aliasing as circular convolution is employed To form the output sequence the first M-1 data points are discarded in each output block and the remaining data are fitted together Overlap-add method In this method the size of the input data block is L Each data block is L points and we append M-1 zeros to compute N point DFT In this no corruption due to aliasing as linear convolution is performed using circular convolution To form the output sequence the last M-1 points from each output block is added to the first M-1 points of the succeeding block	**C312.2**	BTL 1
40	**What are the differences between DIF and DIT algorithms?** Differences: 1)The input is bit reversed while the output is in natural order for DIT, whereas for DIF the output is bit reversed while the input is in natural order.	**C312.2**	BTL 1

	2)The DIF butterfly is slightly different from the DIT butterfly, the difference being that the complex multiplication takes place after the add-subtract operation in DIF.		
41	**What are the similarities between DIF and DIT algorithms**? Similarities: Both algorithms require same number of operations to compute the DFT.Both algorithms can be done in place and both need to perform bit reversal at some place during the computation.	**C312.2**	BTL 1
42	**What is FFT?** The Fast Fourier Transform is an algorithm used to compute the DFT. It makes use of the symmetry and periodicity properties of twiddle factor to effectively reduce the DFT computation time. It is based on the fundamental principle of decomposing the computation of DFT of a sequence of length N into successively smaller DFTs.	**C312.2**	BTL 1
43	**state the linearity properties of Z-transform.** if x1(n)↔X1(Z) and x2(n)↔X2(Z) then Z a1x1(n)+a2x2(n)↔a1X1(Z)+a2X2(Z)	**C312.2**	BTL 1
44	**state the Time shifting properties of Z-transform** if x(n)↔X(Z) then Z x(n-k)↔Z-KX(Z)	**C312.2**	BTL 1
45	**state the Scaling in Z-domain properties of Z-transform** if x(n)↔X(Z) then anx(n)↔X(a-1Z)	**C312.2**	BTL 1
46	**state the Differtiation in Z domain properties of Z-transform** nx(n)↔-Zdz X(Z)	**C312.2**	BTL 1
47	**state the correlation properties of Z-transform** if x1(n)↔X1(Z) and x2(n)↔X2(Z) then ∞ Z rx1x2(l=∑x1(n) x2(nl)↔Rx1x2(Z)=X1(Z) .X2(Z-1) n=-∞	**C312.2**	BTL 1

48	**state the convolution of two sequences properties of Z-transform** if x1(n)↔X1(Z) and x2(n)↔x2(Z) then x1(n)*x2(n)↔X(Z)=X1(Z).X2(Z)	**C312.2**	BTL 1
49	**How many multiplications are required to compute N point DFT using redix-4 FFT?** The number of multiplications required to compute N point DFT using radix-2 FFT are 3N log2 N	**C312.2**	BTL 2
50	**Give any two application of DCT**? **<u>Nov/Dec 2018</u>.** The DFT is used for spectral analysis of signals using a digital computer. The DFT is used to perform filtering operations on signals using digital computer The DCT is used for spectral analysis of signals using a digital computer. The DCT is used to perform filtering operations on signals using digital computer	**C312.2**	BTL 2

PART-B

Q. NO.	**QUESTIONS**	**CO**	**Bloom's Level**
1	**Determine the DFT of the sequence x(n) = {1,1,-2,-2} <u>MAY/JUNE 2016</u>,** **ii) Determine the response of LTI system by radix -2 DIT FFT.** **<u>Nov/Dec2006, APRIL/MAY2015</u>** Ans: i) X(K) = { 0, -1-j,6,-1+j} ii) Ref Pg.No 320-328 , DSP by proakis	**C312.2**	BTL 5
2	**Derive the equation for Decimation – in time algorithm for FFT**. **ii) How do you perform linear filtering by FFT using Save –add method? <u>(Nov/Dec 2010, Nov/Dec 2006 & April /May 2008 & Nov/Dec 2008) April/May 2017</u>** Ans: i) Ref Pg.No 320-328 , DSP by proakis . ii) Ref Pg.No 369, DSP by proakis	**C312.2**	BTL 3
3	**.i) Prove the following properties of DFT when H(K) is the DFT of an N-point sequence h(n).**	**C312.2**	BTL 4

	1. H(K) is real and even when h(n) is real and even. **H(K) is imaginary and odd when h(n) is real and odd** **ii) Compute the DFT of $x(n) = e^{-0.5n}$, 0≤ n≤ 5. May/June 2007,May/June 2013** Ans: i) Ref Pg.No 309, DSP by proakis		
4	**i) From first principles obtain the signal flow graph for Computing 8-point using radix -2 DIF –FFT algorithm. ii) Using the above signal flow graph compute DFT of x(n) = cos (nπ/4) ,0 ≤ n ≤ 7. (May/June 2007 & Nov/Dec 2007 & Nov/Dec 2008 ,Apr/May 2011,May /June 2012)** Ans: i) Ref Pg.No 334-340, DSP by proakis . ii) X(K) = {0, 3, 0, 2.7-j0.7, 0, 1, 0, 1.293-j0.7}	**C312.2**	BTL 3
5	**Two finite duration sequence are given by x(n) = sin (nπ/2) for n = 0,1,2,3 h(n) = 2 n for n = 0,1,2,3 Determine circular convolution using FT &IDFT method. Nov/Dec 2007** Ans: X(K) = {0, -2j, 0, 2j} H(K) = {15, -3+6j, -5, -3-6j} y(n) = {6, -3, -6, 3}	**C312.2**	BTL 4
6	**Discuss in detail the important properties of the DFT. Nov/Dec 2018. ii) Find the 4-point DFT of the sequence x(n) = cos (nπ/4) iii) Compute an 8-point DFT using DIF FFT radix -2 algorithm. x(n) = { 1,2,3,4,4,3,2,1} (May/June 2012 , Nov /Dec 2010 ,April /May 2008)** Ans: i)Ref Pg.No 308-311, DSP by proakis . ii) X(K) = {1, 1-j1.414, 1, 1+j1.414} iii) X(K) = {20,-5.8-j2.4, 0, 0.17-j0.414, 0, -0.17+j0.414, 0, -5.82+j2.414}.	**C312.2**	BTL 6
7	**Determine eight point DFT of the following sequences using radix2 DIFFFT** **algorithm x(n)={1,-1,-1,-1,1,1,1,-1}.May/June 2013 MAY/JUNE 2016, Nov/DEC 2018** Refer notes	**C312.2**	BTL 5
8	**Find eight point DFT of the following sequences using radix2 DITFFT algorithm x(n)={1,-1,1,-1,0,0,0,0,0}.May/June 2014, APRIL/MAY2015 Nov/Dec2017**	**C312.2**	BTL 5

	Refer notes		
9	**Using radix 2 DIT-FFT algorithm ,determine DFT of the given sequence for N=8 x(n)=n for 0<n<7 Apr/May 2017** Refer notes	**C312.2**	BTL 5
10	**Determine the response of LTI system when input x(n)= {-1,1,2,1} and impulse response h(n)={-1,1,-1,1] by Radix 2 DIT FFT April/May 2017**	**C312.2**	BTL 4
11	**Given x(n)={1 2 3 4 }=h(n). circularly convolve x(n) and h(n) using DFT and IDFT computations. Nov/Dec2017.Nov/DEC 2018** Refer notes	**C312.2**	BTL 5
12	**Given x(n)={1 2 -1 ,2,2,-1,2,1}, Compute 8 point DFT using Decimation in time algorithms. Nov/Dec2017.** Refer notes	**C312.2**	BTL 5
13	**Explain the filtering methods based on DFT and FFT. April/May 2017** Refer notes	**C312.2**	BTL 2
14	**Determine the response of LTI system when input sequence x(n)={-1,1,2,1} and impulse response h(n)={-1,1,-1,1) by Radix2 DITFFT.** Refer notes	**C312.2**	BTL 6
15.	**Starting from the key equation of DFT ,with necessary equation explain DIT-FFt algorithm. April/May 2018** Refer notes	**C312.2**	BTL 2
16.	**Determine IDFT of X(K)={6,-2,-2j,2,-2+2j} using DIT algorithm. Nov/DEC 2018**		

UNIT III PART A

<table>
<tr><th>Q. No.</th><th>Questions</th><th>CO</th><th>Bloom's level</th></tr>
<tr><td>1</td><td>List any two properties of Butterworth filter. Nov/Dec 2006, Nov/Dec 2011, Apr/May 2011
Properties of Butterworth: The Butterworth filters are all pole design. The filter order N completely specifies the filter The magnitude is maximally flat at the origin. The magnitude is monotonically decreasing function of ohm.</td><td>C312.3</td><td></td></tr>
<tr><td>2</td><td>Find the digital transfer function H(Z) by using impulse invariant method for the analog transfer function H(S) = 1/ (S+2).Assume T=0.5sec May /June 2007 &Nov/Dec 2007
H(S) = 1/ (S+2). H(Z) = 1/[1-e-1 Z-1] (Z) = 1/ [1-0.368Z-1]</td><td>C312.3</td><td>BTL 1</td></tr>
<tr><td>3</td><td>What is the relationship between analog and digital frequency in impulse invariant transformation? April/May 2008
Digital Frequency: $\omega = \Omega T$ Ω = analog frequency T= Sampling interval</td><td>C312.3</td><td>BTL 1</td></tr>
<tr><td>4</td><td>What is Prewarping Or Warping effect ? Why is it needed? Nov/Dec 2008 ,May /Jun 2012 MAY/JUNE 2016, April/May 2018
In IIR design using bilinear transformation the conversion of specified digital frequencies to analog frequencies is called Pre-warping. The Pre-Warping is necessary to eliminate the effect of warping on amplitude response</td><td>C312.3</td><td>BTL 1</td></tr>
<tr><td>5</td><td>Compare FIR & IIR filter.
<table>
<tr><th>S. No</th><th>FIR filter</th><th>IIR filter</th></tr>
<tr><td>1.</td><td>Only N samples of impulse response are considered.</td><td>All the infinite samples of impulse considered.</td></tr>
<tr><td>2.</td><td>Linear phase characteristics can be achieved</td><td>Linear phase characteristics cannot be achieved</td></tr>
</table>
</td><td>C312.3</td><td>BTL 4</td></tr>
<tr><td>6</td><td>Define Frequency warping. Nov/Dec 2011 April/May 2017
The non linear relationship between analog and digital frequencies</td><td>C312.3</td><td>BTL 1</td></tr>
</table>

	introduced frequency distortion which is called as frequency warping.		
7	**1. Compare Butterworth with chebychev filters.(May/June 2012)** The magnitude response of Butterworth filter decreases monotonically as the frequency Ω increases from 0 to ∞, whereas the magnitude response of the chebychev filter exhibits ripples in the pass band or stop band according to the type. The transition band is more in Butterworth filter when compared to chebychev filter. The poles of the Butterworth filter lie on a circle, whereas the poles of the chebychev filter lie on an ellipse..For the same specification, the number of poles in Butterworth are more when compared to the chebychev filter ie. The order of the chebychev filter is less than that of Butterworth. This is a great advantage because less number of discrete components will be necessary to construct the filter.	**C312.3**	BTL 4
8	**What are the properties of impulse invarient transformation. MAY/JUNE 2014** It is many to one mapping. The relocation between analog and digital frequency is linear. To prevent the problem of aliasing ,the analog filter should be band limited. The magnitude and phase response of analog filter can be preserved by choosing low sampling time or high sampling frequency	**C312.3**	BTL 1
9	**Draw the direct form structure of IIR filter. MAY/JUNE 2014, APRIL/MAY2015** **Refer notes**	**C312.3**	BTL 1
10	**What is meant by bilinear transformation method of designing IIR filter. APRIL/MAY2015** This transformation is a one to one mapping from the S domain to Z domain	**C312.3**	BTL 1
11	**Write the advantages of bilinear transformation.** 1.There is no aliasing.2.The effect of warping on amplitude response can be eliminated by prewarping the analog filter.3.It is one to one mapping	**C312.3**	BTL 1
12	**Write the disadvantages of bilinear transformation.** The non linear relationship between the analog and digital frequencies introduce frequency distortion which is called as frequency warping. Using BLT a linear phase analog filter cannot be transformed to linear phase digital filter.	**C312.3**	BTL 1

13	**Write the Properties of Chebychev filter: <u>May/June 2013 MAY/JUNE 2016,</u>** The magnitude response of the filter exhibits ripples in the pass band or stop bandThe pole of the filter lies on an ellipse.	**C312.3**	BTL 1
14	**Write the structural realization of FIR filter.** 1.Direct form I 2.Direct form II3.Cascade form4.Parallel form	**C312.3**	BTL 1
15	**Write the design types of butterworth filter.** 1,Bilinear transformation 2.Impulse Invariant Method	**C312.3**	BTL 1
16	**Write the design types of Chebychev filter**. 1.Bilinear transformation 2.Impulse Invariant Method	**C312.3**	BTL 1
17	**Define IIR filter.** All the infinite samples of impulse response are considered in IIR filter.	**C312.3**	BTL 1
18	**What is analog frequency transformation**? Using analog frequency transformation the following filters can be designed 1. low pass filter of another Ω c 2.High pass filter with Ω c 3.band pass filter with centre frequency Ω 0 and quality factor Q 4..band pass filter with centre frequency Ω 0 and quality factor Q	**C312.3**	BTL 1
19	**Give the transform relation for converting low pass to band pass in digital domain. <u>(Apr 2004</u>)** Low pass with cut – off frequency ΏC to band –pass with lower cut-off frequency Ώ1 and higher cut-off frequency Ώ2: S ------------ ΏC (s2 + Ώ1 Ώ2) / s (Ώ2 - Ώ1) The system function of the high pass filter is then H(s) = Hp { ΏC (s2 + Ώ1 Ώ2) / s (Ώ2 - Ώ1)}	**C312.3**	BTL 1
20	**Write the magnitude function of Butterworth filter. What is the effect of varying order of N on magnitude and phase response<u>?(Nov 2005</u>)** |H(jΏ)|2 = 1 / [1 + (Ώ/ΩC)2N] where N= 1,2,	**C312.3**	BTL 1
21	**What is the relation between analog and digital frequency in impulse invariant transformation? <u>(April 2008</u>)** ΩT= ω	**C312.3**	BTL 1
22	**Find the digital transfer function H(z) by using impulse invariant method for the analog transfer function H(s) = 1/ (s+2). Assume**	**C312.3**	BTL 1

	T=0.1 sec. (Nov 2007) H(Z)=1/(1-e(-p1*T)z-1) H(Z)=1/(1-e(-0.2)z-1)		
23	**State the condition for a digital filter to be causal and stable. (May 2007)** The response of the causal system to an input does not depend on future values of that input, but depends only on the present and/or past values of the input. A filter is said to be stable, bounded-input bounded output stable, if every bounded input produces a bounded output. A bounded signal has amplitude that remains finite.	**C312.3**	BTL 1
24	**Mention any two procedures for digitizing the transfer function of an analog filter. Nov 2006)or Write the transformation which is used for conversion of analog domain to digital domain by using bilinear transformation. Nov/DEC 2018** Impulse Invariant Technique Bilinear Transform Technique	**C312.3**	BTL 2
25	**Give the equation for the order N, major, minor and axis of an ellipse in case of Chebyshev filter. (Nov 2005)** N ≥ cosh-1 (λ/ε) / cosh-1(ΏS/ ΏP) Where λ = √ (100.1αs – 1) ε = √ (100.1αp – 1)	**C312.3**	BTL 1
26	**What are the advantages of bilinear transformation? (May 2006)** **Advantages:** Many to one mapping .linear frequency relationship between analog and its transformed digital frequency is simpler.	**C312.3**	BTL 1
27	**Name the different design techniques for designing IIR filter. (Nov/Dec 2009 [R2001])** Chebyshev's Filter Butterworth Filter	**C312.3**	BTL 1
28	**Using approximation of derivatives convert the following analog filter into digital filter H(s) = 1/(S+1) (Nov/Dec 2009 [R2001])** Refer notes	**C312.3**	BTL 1
29	**What are the limitations of impulse invariant mapping technique? (Apr2004, Nov/Dec 2009** The impulse invariance technique is appropriate only for band limited filter like low pass filter. Impulse invariance design for high pass or band stop continuous-time filters, require additional band limiting to avoid severe aliasing distortion, if impulse designed is used. Thus this method is not preferred in the design of IIR filters other than low-pass filters.	**C312.3**	BTL 1
30	**Find the equivalent digital transfer function H (z) by using impulse invariant method for the analog transfer function H(s) = 1/(S+2).**	**C312.3**	BTL 1

	Assume T=0.5sec.H(s) = 1/s+2 Nov/DEC 2017 The system function of the digital filter is obtained byH (z) = 1/ (1-e-2Tz-1) Since T=o.5 sec H (z) = 1/ (1-.067Z^{-1}		
31	**What are the different types of filters based on impulse response?** Based on impulse response the filters are of two types 1. IIR filter 2. FIR filter The IIR filters are of recursive type, whereby the present output sample depends on the present input, past input samples and output samples. The FIR filters are of non recursive type, whereby the present output sample depends on the present input sample and previous input samples	**C312.3**	BTL 1
32	**What are the different types of filters based on frequency response?** Based on frequency response the filters can be classified as 1. Lowpass filter 2. Highpass filter 3. Bandpass filter 4. Bandreject filter	**C312.3**	BTL 1
33	**What is the transfer function of H(S) for a normalized Butterworth filter** **Nov/Dec2017** **Refer notes**	**C312.3**	BTL 1
34	**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 $n\leq 0$	**C312.3**	BTL 1
35	**State the advantage of direct form II structure over direct form I structure**. In direct form II structure, the number of memory locations required is less than that of direct form I structure.	**C312.3**	BTL 1
36	**How one can design digital filters from analog filters?** • Map the desired digital filter specifications into those for an equivalent	**C312.3**	BTL 2

	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.		
37	**Mention the methods for converting analog into digital IIR filter. April/May 2018**. The two important procedures for digitizing the transfer function of an analog filter are • Impulse invariance method. • Bilinear transformation method.	**C312.3**	BTL 1
38	**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.	**C312.3**	BTL 1
39	**What is the mapping procedure between S-plane & Z-plane in the method of mapping differentials?** The mapping procedure between S-plane & Z-plane in the method of mapping of differentials is given by $H(Z) =H(S)\|S=(1-Z-1)/T$.	**C312.3**	BTL 1
40	**What is meant by impulse invariant method of designing IIR filter?** In this method of digitizing an analog filter, the impulse response of resulting digital filter is a sampled version of the impulse response of the analog filter. The transfer function of analog filter in partial fraction form,	**C312.3**	BTL 1
41	**Give the bilinear transform equation between S-plane & Z-plane.** $S=2/T(1-Z-1/1+Z-1)$	**C312.3**	BTL 1
42	**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.	**C312.3**	BTL 1

	The mapping from the S-plane to the Z-plane is in bilinear transformation is S=2/T(1-Z-1/1+Z-1)		
43	**What are the properties 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, and vice-versa. • The j Ω-axis maps on to the unit circle |z|=1,the left half of the s-plane maps to the interior of the unit circle |z|=1 and the half of the s-plane maps on to the exterior of the unit circle |z|=1.	**C312.3**	BTL 1
44	**What are the advantages of bilinear transformation?** Advantages: • The bilinear transformation provides one-to-one mapping. • Stable continuous systems can be mapped into realizable, stable digi systems. • There is no aliasing.	**C312.3**	BTL 1
45	**What are the disadvantages of bilinear transformation?** Disadvantage: • The mapping is highly non-linear producing frequency, compression at high frequencies. • Neither the impulse response nor the phase response of the analog filter is preserved in a digital filter obtained by bilinear transformation	**C312.3**	BTL 1
46	**What is the advantage of cascade realization?** Quantization errors can be minimized if we realize an LTI system in casca form.	**C312.3**	BTL 1
47	**What are characteristics between S-plane & Z-plane** **S-plane & Z-plane** mapping has the following characteristics • The left half of S-plane maps inside a circle of radius ½ centered at Z= ½ in the Z-plane. • The right half of S-plane maps into the region outside the circle of radius ½ in the Z-plane. • The j Ω-axis maps onto the perimeter of the circle of radius ½ in the Z-plane	**C312.3**	BTL 1

48	**What are disadvantages of Impulse invariant transformation? (May 2006)or why impulse invariant transformation is not suitable for the design of high pass filter. Nov/Dec 2018.** **Disadvantage**: Aliasing	**C312.3**	BTL 1
49	**What is the disadvantage of cascade realization?** circuit complexity is more	**C312.3**	BTL 1
50	**What are realization of FIR filter?** 1.Direct form I 2.Direct form II3.Cascade form4.Parallel form	**C312.3**	BTL 1

PART-B

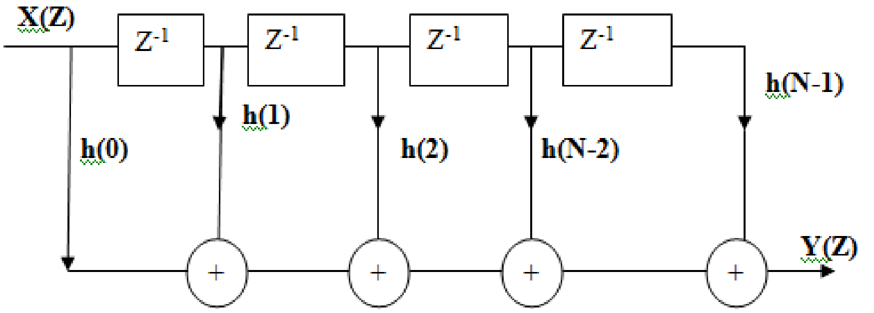
Q. NO.	QUESTIONS	CO	Bloom's Level
1	**Design a digital Butterworth filter satisfying the constraints using bilinear transformations. APRIL/MAY2015** $\mathbf{0.707 \le \lvert H(\omega)\rvert \le 1.0 \;;\; 0 \le \omega \le \pi/2}$ $\mathbf{\lvert H(\omega)\rvert \le 0.2 \;;\; 3\pi/4 \le \omega \le \pi.}$ Ans: Ref Pg.No 435-437, DSP by proakis	**C312.3**	BTL 6
2	**Design a digital Butterworth filter satisfying the constraints** $\mathbf{0.8 \le \lvert H(\omega)\rvert \le 1.0 \;;\; 0 \le \omega \le \pi/4}$ $\mathbf{\lvert H(\omega)\rvert \le 0.2 \;;\; \pi/2 \le \omega \le \pi.}$ **Apply Bilinear transformation method. May/June2007 & Nov/Dec 2008 MAY/JUNE 2016, April/May 2017,Nov/Dec2018** Ans: Ref: Pg.No: 359-362, DSP by proakis.	**C312.3**	BTL 6
	i)Design a digital BUTTERWORTH filter that satisfies the following constraint using BILINEAR Transformation. Assume T = 1 sec. $\mathbf{0.9 \le \lvert H(\omega)\rvert \le 1 \;;\; 0 \le \omega \le \pi/2}$ $\mathbf{\lvert H(\omega)\rvert \le 0.2 \;;\; (3\pi/4) \le \omega \le \pi}$ **ii) Determine the magnitude response of the FIR filter (M=11) and show that Phase and group delay are constant iii) The desired frequency response of a low pass**		

3	**filter is given by Hd(ω) ={ e –j3ω ; -3π/4 ≤ ω ≤ 3π/4 0 ; otherwise. Determine H(ejω) for M= 7using HAMMING window.** **iv) For the analog transfer function H(S) = 1/ (S+1)(S+2) . Determine H(Z) using impulse invariant technique. <u>April /May 2008 April/May 2017</u>** Ans: i) Ref Pg.No 437-439, DSP by proakis. ii) Ref Pg.No 383-384, DSP by proakis. iii) Ref Pg.No 400-401, DSP by proakis. iv) Ref Pg.No 426, DSP by proakis.	**C312.3**	BTL 6
4	**Obtain the direct form-I, direct form –II , cascade form and parallel form realization of the following system function.** **Y(n)=-0.1 y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)** **<u>Nov/Dec 2010 , Nov/Dec 2011) MAY/JUNE 2014, APRIL/MAY2015 April/May 2017</u>** Refer notes	**C312.3**	BTL 5
5	**Realize the following FIR system with difference equation** . y(n)=3/4 y(n-1)-1/8y(n-2)+x(n)+1/3x(n-1) in direct form I. **<u>MAY/JUNE 2014</u>**	**C312.3**	BTL 6
6	**Design a digital chebyshev filter satisfying the constraints** **0.75 ≤ | H(ω)| ≤ 1.0 ; 0 ≤ ω ≤ π/2** **| H(ω)| ≤ 0.2 ; 3π/4 ≤ ω ≤ π.** **Apply Bilinear transformation method. <u>MAY/JUNE 2014</u>.** **Refer Notes**	**C312.3**	BTL 6
7	**Explain with necessary equations the approximations derivatives method for converting an analog filter into a digital filter. <u>Nov/Dec 2017.</u>** Refer Notes	**C312.3**	BTL 2
8	**Using bilinear transformation design a lowpass filter monotonic in passband with -3.01 db cutoff frequency of 0.4 π rad and magnitude down atleast by 15 db at 0.75 π rad <u>Nov/Dec 2017</u>** Refer Notes	**C312.3**	BTL 6
9	**The specifications of desired low pass filter is** **0.8 ≤ | H(ω)| ≤ 1.0 ; 0 ≤ ω ≤ 0.2π** **| H(ω)| ≤ 0.2 ; 0.6π ≤ ω ≤ π.Design a Chebyshev digital filter using impulse invariant Transformation. <u>Apr/May</u>**	**C312.3**	BTL 6

	2017.Nov/Dec2018 Refer notes		
10	**Determine the system function of the IIR digital filter for the analog transfer function $H(S)=10/S2+7S+10$ with $T=0.2$ sec using impulse invariant method. Apr/May 2017** Refer notes	**C312.3**	BTL 6
11	**Design a digital filter which exhibits equiripple behavior only either in passband or stopband and monotonic satisfying the constraints** $0.8 \le \lvert H(\omega)\rvert \le 1.0$ **;** $0 \le \omega \le 0.2\pi$ $\lvert H(\omega)\rvert \le 0.2$ **;** $0.6\pi \le \omega \le \pi$. **Using Bilinear transformation method. April/May 2018.** Refer notes	**C312.3**	BTL 6
12	**Convert the analog filter with transfer function $H(s)=2/(s+1)(s+2)$ into digital filter using Impulse Invarient method. April/May 2018** Refer notes	**C312.3**	BTL 4
13	**The specifications of desired low pass filter is** $0.79 \le \lvert H(\omega)\rvert \le 1.0$ **;** $0 \le \omega \le 0.2\pi$ $\lvert H(\omega)\rvert \le 0.2$ **;** $0.6\pi \le \omega \le \pi$.**Design a Chebyshev digital filter using Bilinear transformation .** Refer notes	**C312.3**	BTL 6
14	**Realize the following FIR system with difference equation** . $y(n)=3/4\, y(n-1)-1/8y(n-2)+x(n)+1/3x(n-1)$ **in direct form II** Refer notes	**C312.3**	BTL 6
15	**Write the design procedure for butterworth filter and Chebeshev filter** Refer notes**.**	**C312.3**	BTL 1

UNIT IV

PART – A

Q. No.	Questions	CO	Bloom's Level
1	**Draw the block diagram representation of a FIR system? Nov/Dec 2006**  	**C312.4**	BTL 1
2	**Show that the h(n) = [-1,0,1] is a linear phase filter. May /June 2007 Nov/Dec 2008** h(n) = [-1,0,1] h(0) = -1 = -h(N-1-n) = -h(3-1-0) = -h(2) h(1) = 0 = -h(N-1-n) = -h(3-1-1) = -h(1 It is a linear phase filter.	**C312.4**	BTL 3
3	**In the design of FIR digital filter, how is Kaiser Window different from other windows? Nov/Dec 2007** In all other windows a trade off exists between ripple ratio and main lobe width. In Kaiser Window both ripple ratio and main lobe width can be varied independently	**C312.4**	BTL 1
4	**What are the merits and demerits of FIR filter? April/May 2008** **Merits** :Linear phase filter. Always Stable **Demerits**: The duration of the impulse response should be large Non integral delay.	**C312.4**	BTL 1
5	**What are the advantages of FIR filter over IIR filter? April/May 2017** They can have an exact linear phase. They are always stable They can be realized efficiently in hardware The design methods are	**C312.4**	BTL 1

<table>
<tr><td></td><td>generally stable.</td><td></td><td></td></tr>
<tr><td>6</td><td>**What is the necessary & sufficient condition of linear phase FIR filter? (May/June 2012)or Write the condition for FIR filter to have linear phase.Nov/Dec2018**
The condition for a linear phase filter is $\alpha = (N-1)/2$ $h(n) = h(N-1-n)$</td><td>**C312.4**</td><td>BTL 1</td></tr>
<tr><td>7</td><td>**What is Gibb’s phenomenon?(Apr/May 2011, May /June 2012),Nov/Dec 2012 April/May 2017**
In Fir filter design using Fourier analysis method for rectangular window method, the infinite duration impulse response is truncated to finite duration impulse response. The abrupt truncation of impulse response introduces a oscillation in the pass band and stop band .This effect is Known as Gibb’s phenomenon</td><td>**C312.4**</td><td>BTL 1</td></tr>
<tr><td>8</td><td>**Compare Rectangular & Hamming window.**
<table>
<tr><td>S.No</td><td>Rectangular Window</td><td>Hamming window.</td></tr>
<tr><td>1.</td><td>The width of the main lobe in window spectrum is $4\pi/N$</td><td>The width of the main lobe in window spectrum is $8\pi/N$</td></tr>
<tr><td>2.</td><td>The maximum side lobe magnitude in window spectrum is -13 dB</td><td>The maximum side lobe magnitude in window spectrum is -41 dB</td></tr>
</table></td><td>**C312.4**</td><td>BTL 2</td></tr>
<tr><td>9</td><td><table>
<tr><td>S.No</td><td>Kaiser Window</td><td>Hamming window.</td></tr>
<tr><td>1.</td><td>The width of the main lobe in window spectrum depends on the value of α and N.</td><td>The width of the main lobe in window spectrum is $8\pi/N$</td></tr>
<tr><td>2.</td><td>The maximum side lobe magnitude with respect to peaK of main lobe is variable using the parameter α.</td><td>The maximum side lobe magnitude in window spectrum is -41 dB</td></tr>
</table></td><td>**C312.4**</td><td>BTL 2</td></tr>
<tr><td>10</td><td>**Compare FIR & IIR filter.**</td><td>**C312.4**</td><td>BTL 2</td></tr>
</table>

<table>
<tr><td></td><td>
<table>
<tr><th>S.No</th><th>FIR filter</th><th>IIR filter</th></tr>
<tr><td>1.</td><td>Only N samples of impulse response are considered.</td><td>All the infinite samples of impulse response are considered.</td></tr>
<tr><td>2.</td><td>Linear phase characteristics can be achieved</td><td>Linear phase characteristics cannot be achieved</td></tr>
</table>
</td><td></td><td></td></tr>
<tr><td>11</td><td>Compare Rectangular Window& Hanning Window.
<table>
<tr><td>S.No</td><td>Rectangular Window</td><td>Hanning Window</td></tr>
<tr><td>1.</td><td>The width of the main lobe in window spectrum is 4π/N</td><td>The width of the main lobe in window spectrum is 8π/N</td></tr>
<tr><td>2.</td><td>The maximum side lobe magnitude in window spectrum is -13 dB</td><td>The maximum side lobe magnitude in window spectrum is -31 dB</td></tr>
</table>
</td><td>C312.4</td><td>BTL 2</td></tr>
<tr><td>12</td><td>Compare Hamming Window& Hanning Window.
<table>
<tr><td>S.No</td><td>Hamming window.</td><td>Hanning Window</td></tr>
<tr><td>1.</td><td>The width of the main lobe in window spectrum is 8π/N</td><td>The width of the main lobe spectrum is 8π/N</td></tr>
<tr><td>2.</td><td>The maximum side lobe magnitude in window spectrum is -41 dB</td><td>The maximum side lobe magnitud spectrum is -31 dB</td></tr>
</table>
</td><td>C312.4</td><td>BTL 2</td></tr>
<tr><td>13</td><td>Compare Hamming Window& BlacKman Window.<u>May/June 2013</u></td><td>C312.4</td><td>BTL 2</td></tr>
</table>

S.No	Hamming window.	BlacKman Window
1.	The width of the main lobe in window spectrum is 8π/N	The width of the main lobe in window spectrum is 12π/N
2.	The maximum side lobe magnitude in window spectrum is -41 dB	The maximum side lobe magnitude in window spectrum is -58 dB

14	**Give the equations for Hamming window and BlacKman window. <u>(Nov/Dec 2010) May/June 2013</u>** • $Hammin\,g Window W_H(n) = \begin{Bmatrix} 0.54 - 0.46Cos(\frac{2\pi n}{N-1}); 0 \le n \le N-1 \\ 0; otherwise \end{Bmatrix}$ $Blackman Window W_B(n) = \begin{Bmatrix} 0.42 - 0.5Cos(\frac{2\pi n}{N-1}) + 0.08Cos(\frac{4\pi n}{N-1}); 0 \le n \le N-1 \\ 0; otherwise \end{Bmatrix}$	**C312.4**	BTL 1
15	**What are the properties of FIR filter<u>?(Apr/May 2011 , Nov/Dec 2011), APRIL/MAY2015 MAY/JUNE 2016,</u>** FIR filters are stable.FIR filters have linear phase. They need higher order filters for the same magnitude response compared to IIR Filters.	**C312.4**	BTL 1
16	**What is the reason that FIR filter is always stable? <u>MAY/JUNE 2014</u>** The phase delay and group delay of a linear phase FIR filter are equal and constant over the frequency band whenever a constant group delay is preferred the impulse response will be in the form of H(n)=-h(N-1-n) and it is anti symmetric about the centre of the impulse response sequence.	**C312.4**	BTL 1
17	**What do you understand by linear phase response in filters? <u>MAY/JUNE 2014, APRIL/MAY2015</u>** The linear phase filter are those in which the phase delay and group delay are constant. The linear phase filter is also called as constant time delay filter.	**C312.4**	BTL 1
18	**Define FIR filter.** Only N samples of impulse response are considered. Linear phase	**C312.4**	BTL 1

	characteristics can be achieved **.**		
19	**What is constant time delay filter?** The linear phase filters are those in which the phase delay and group delay are constant. The linear phase filter is also called as constant time delay filter.	**C312.4**	BTL 1
20	**What is group delay and phase delay.** Filters can have linear or nonlinear phase depending upon the delay function namely phase delay and group delay .phase delay=-o(w)/w group delay=-do(w)/d(w)	**C312.4**	BTL 1
21	**Show that the filter with h (n) = [-1, 0, 1] is a linear phase filter. <u>(Nov 2008,May 2007)</u>** $\mathbf{H(e^{jw})} = \sum_{n=-\infty}^{\infty} h(n)e^{-jnw}$ $= -1 + e^{-j2w}$ $= e^{-w}[e^{-w}- e^{w}]$ $= e^{-w}(-2jsinw)$ $=-2j\, e^{-w}sinw$ We can find $\theta(w)=-w$ Which is proportional to w. Hence the filter h(n) is a linear phase filter	**C312.4**	BTL 1
22	**What is linear phase? What is the condition to be satisfied by the impulse response in order to have a linear phase? <u>(Apr 2005 & Nov 2003)</u>** For a filter to have linear phase the phase response $\theta(w)\ \alpha\ w$ is the angular frequency. The linear phase filter does not alter the shape of the signal. The necessary and sufficient condition for a filter to have linear phase. $h(n) = \pm h(N-1-n)$; $0 \leq n \leq N-1$	**C312.4**	BTL 1
23	**Give the Kaiser Window function. Apr 2004)** The Kaiser Window function is given by	**C312.4**	BTL 1

	$W_K(n) = I_0(\beta) / I_0(\alpha)$, for $\lvert n\rvert \le (M-1)/2$ Where α is an independent variable determined by Kaiser. $B = \alpha[1 – (2n/M-1)^2]$		
24	**State the expression for Hamming window. (Nov/Dec 2009[R2001])** $W_H(n)= 0.54+0.46\cos(2\pi n/N-1)$ for $–(N-1)/2 \le n\le (N-1)/2$ $= 0$ Otherwise	**C312.4**	BTL 1
25	**In the design of FIR digital filters, how is Kaiser window different from other windows? (Nov 2007)** It provides flexibility for the designer to select the side lobe level and N.It has the attractive property that the side lobe level can be varied continuously from the low value in the BlacKman window to the high value in the rectangular window	**C312.4**	BTL 1
26	**What are the desirable and undesirable features of FIR Filters? (May2006)**The width of the main lobe should be small and it should contain as much of total energy as possible. The side lobes should decrease in energy rapidly as w tends to π	**C312.4**	BTL 1
27	**List the characteristics of FIR filters designed using window functions. (Nov 2004)** The Fourier transform of the window function $W(e^{jw})$ should have a small width of main lobe containing as much of the total energy as possible the Fourier transform of the window function $W(e^{jw})$ should have side lobes that decrease in energy rapidly as w to π. Some of the most frequently used window functions are described in the following sections.	**C312.4**	BTL 1
28	**Define Hanning and BlacKman window functions. (May 2006)** The window function of a causal Hanning window is given by $W_{Hann}(n) = 0.5 – 0.5\cos2\pi n/ (M-1), 0\le n\le M-1$ The width of the main lobe is approximately 8π/M and thee peaK of the first side lobe is at -32dB**.** The window function of a causal BlacKman window is expressed by $W_B(n) = 0.42 – 0.5 \cos2\pi n/ (M-1) +0.08 \cos4\pi n/(M-1), 0\le n\le M-1$ The width of the main lobe is approximately	**C312.4**	BTL 1

	$12\pi/M$ and the peaK of the first side lobe is at -58dB.		
29	**Mention the necessary and sufficient condition for linear phase characteristics in FIR filter. <u>(Nov 2005)</u>** The necessary and sufficient conditions is that the phase function should be linear function w, which in turn requires constant phase delay (or) constant phase and group delay i.e., $Q(w) \alpha w$ $Q(w) = - \alpha w$ $-\pi \leq w \leq \pi$	**C312.4**	BTL 1
30	**List the characteristics of FIR filters designed using window functions. (Nov 2004)<u> MAY/JUNE 2016,</u>** The Fourier transform of the window function $W(e^{jw})$ should have a small width of main lobe containing as much of the total energy as possible The Fourier transform of the window function $W(e^{jw})$ should have side lobes that decrease in energy rapidly as w to π. Some of the most frequently used window functions are described in the following sections.	**C312.4**	BTL 1
31	**what are various windows used for designing FIR filters**.**<u>Nov/Dec 2017</u>** Hamming ,Hanning, Rectangular	**C312.4**	BTL 1
32	**What are the design techniques of designing FIR filters?** There are three well known methods for designing FIR filters with linear phase .They are (1.)Window method (2.)Frequency sampling method (3.)Optimal or minimax design.	**C312.4**	BTL 1
33	**List the steps involved in the design of FIR filters using windows.** 1.For the desired frequency response Hd(w), find the impulse response hd(n) using Equation πhd(n)=1/2π∫ Hd(w)ejwndw 2.Multiply the infinite impulse response with a chosen window sequence w(n) of length N to obtain filter coefficients h(n),i.e., h(n)= hd(n)w(n) for |n|≤(N-1)/2 = 0 otherwise	**C312.4**	BTL 1
34	**Find the transfer function of the realizable filter** (N-1)/2 H(z)=z-(N-1)/2 [h(0)+∑ h(n)(zn+z-n)] n=0	**C312.4**	BTL 1
35	**What are the desirable characteristics of the window function?** The desirable characteristics of the window are 1.The central lobe of the frequency response of the window should contain	**C312.4**	BTL 1

	most of the energy and should be narrow. 2.The highest side lobe level of the frequency response should be small. 3.The side lobes of the frequency response should decrease in energy rapidly as ω tends to π .		
36	**Give the equations specifying the following Rectangulawindows.** The equation for Rectangular window is given by $W(n)= 1\ 0 \leq n \leq M-1$ 0 otherwis	**C312.4**	BTL 1
37	**Give the window function of Hamming window. Nov/Dec 2018.** Hamming window: The equation for Hamming window is given by $WH(n)= 0.54-0.46 \cos 2\pi n/M-1\ 0 \leq n \leq M-1$ 0 otherwise	**C312.4**	BTL 1
38	**Give the equations specifying the following Hanning window:.** Hanning window: The equation for Hanning window is given by $WHn(n)= 0.5[1- \cos 2\pi n/M-1\ ]\ 0 \leq n \leq M-1$ 0 otherwise	**C312.4**	BTL 1
39	**Give the equations specifying the following Bartlett window:** Bartlett window: The equation for Bartlett window is given by $WT(n)= 1-2\|n-(M-1)/2\|\ 0 \leq n \leq M-1$ M-1 0 otherwise	**C312.4**	BTL 1
40	**Give the equations specifying the following window** Kaiser window: The equation for Kaiser window is given by $Wk(n)= Io[\alpha\sqrt{}1-(2n/N-1)2]$ for $\|n\| \leq N-1$ $Io(\alpha)\ 2$ 0 otherwise where α is an independent parameter.	**C312.4**	BTL 1
41	**Give the impulse responses of an FIR filter h(n)=1,2,3,1,3,2 ,1 .Is it a**	**C312.4**	BTL 1

	linear phase FIR filter.? Justify your answer.Nov/Dec2017 **Refer notes**		
42	**What is the principle of designing FIR filter using frequency sampling method?** In frequency sampling method the desired magnitude response is sampled and a linear phase response is specified .The samples of desired frequency response are identified as DFT coefficients. The filter coefficients are then determined as the IDFT of this set of samples.	**C312.4**	BTL 1
43	**For what type of filters frequency sampling method is suitable?** Frequency sampling method is attractive for narrow band frequency selective filters where only a few of the samples of the frequency response are non zero.	**C312.4**	BTL 2
44	**When cascade form realization** is preferred in FIR filters? The cascade form realization is preferred when complex zeros with absolute magnitude is less than one.	**C312.4**	BTL 1
45	**Compare Hanning window and Hamming window .April.May 2018** Hamming window: The equation for Hamming window is given by $WH(n)= 0.54-0.46 \cos 2\pi n/M-1 \quad 0 \le n \le M-1$ 0 otherwise Hanning window: The equation for Hanning window is given by $WHn(n)= 0.5[1- \cos 2\pi n/M-1] \quad 0 \le n \le M-1$ 0 otherwise	**C312.4**	BTL 2
46	**What is linear phase FIR filter.? April.May 2018.** . The linear phase filter does not alter the shape of the signal. The necessary and sufficient condition for a filter to have linear phase.$h(n) = \pm h(N-1-n)$; $0 \le n \le N-1$	**C312.4**	BTL 2
47	**What are the demerits of FIR filter? April/May 2008** **Demerits**: The duration of the impulse response should be large Non integral delay.	**C312.4**	BTL 1

48	**What is the necessary condition of linear phase FIR filter?** The condition for a linear phase filter is $\alpha = (N-1)/2$ $h(n) = h(N-1-n)$	**C312.4**	BTL 1
49	**What is the sufficient condition of linear phase FIR filter?** The condition for a linear phase filter is $h(n) = h(N-1-n)$	**C312.4**	BTL 1
50	**Write the different windowing techniques used in designing of FIR filter.** Hamming , Hanning, Rectangular	**C312.4**	BTL 1

PART – B

Q. NO.	QUESTIONS	CO	Bloom's Level
1	**Design a high pass filter hamming window by taking 9 samples of w(n) and with a cutoff frequency of 1.2 radians/sec. Nov/Dec 2006 Nov/Dec 2018** Ans: Ref: Pg.No: 298-301, DSP by proakis	**C312.4**	BTL 6
2	**Describe the design of FIR filter using frequency sampling technique. MAY/JUNE 2016,** **ii) The desired frequency response of a low pass filter is given by** **$H_d(\omega) =\{ e^{-j2\omega}$; $-\pi/4 \leq \omega \leq \pi/4$ 0 ; Otherwise. Obtain the filter coefficient, h(n) using RECTANGUAR window** **define by $W(n) = \{ 1; 0 \leq n \leq 4$; otherwise. Nov/Dec 2007** Ans: a) Ref Pg.No 389-391, DSP by proakis b) Ref Pg.No 399, DSP by proakis.	**C312.4**	BTL 5
3	**Design a band pass filter to pass frequencies in the range 1to2radians/sec using Hanning window, with N=5. Nov/Dec 2006** Ans: Ref: Pg.No: 301, DSP by proakis..	**C312.4**	BTL 6
4	**Design an ideal band reject filter using hamming window for the given frequency response.Assume N=11**	**C312.4**	BTL 6

	$H_d(e^{j\omega})=1$; w< π/3 and w>2 π/3 = 0: otherwise **MAY/JUNE 2014** Refer notes		
5	**Design an FIR filter for the ideal frequency response response using Hamming window with N=7$H_d(e^{j\omega})$= $e^{-j3\omega}$; - π/8 <w< π/8** **0 ; π/8<w< π MAY/JUNE 2014 Refer notes**	**C312.4**	BTL 6
6	**Write the design procedures of FIR filter using frequency sampling method. APRIL/MAY2015.** Refer Notes.	**C312.4**	BTL 5
7	**Design an ideal differentiator with frequency response.** **H(e^{jw})=jw;- -π ≤ ω ≤ π using hamming window with N=7** Refer notes	**C312.4**	BTL 6
8	**The desired frequency response of a low pass filter is given by** **$H_d(\omega)$ ={ $e^{-j2\omega}$; -π/4 ≤ ω ≤ π/4** **0 ; Otherwise. Refer Notes** **Obtain the filter coefficient, h(n) using Hamming window define by W(n) = { 1; 0 ≤ n ≤ 4 0; otherwise. Nov/Dec 2007 April/May 2017 Nov/Dec 2017 Nov/Dec 2018** Ans: Refer Notes.	**C312.4**	BTL 5
9	**Determine the filter coeffcients of h(n) of M=15 obtained by sampling and its frequency response is** H(2 **πK/15)=1 for K=01,2,3,4 K=5 K=6,7 April/May** 2017 **Refer Notes**	**C312.4**	BTL 5
10	**Given H(Z) H(z)=0.5+0.25 z-1+0.75z-1+z-3+0.75z-4+0.25z-5+0.5z-6)Draw the linear phase realization and direct form realization and compare both the structures. Nov/Dec 2017 Refer Notes**	**C312.4**	BTL 4
11	**Design an FIR filter for the ideal frequency response using Hamming window with N=7** **$H_d(\omega)$ ={ $e^{-j2\omega}$; -π/8 ≤ ω ≤ π/8** **0 ; Otherwise. Apr/May 2017 Refer Notes**	**C312.4**	BTL6
12	**Determine the filter coefficient of h(n)of length M=15 .obtained by**	**C312.4**	BTL 5

	sampling and its frequency response as **H(2 πK/15)=1 ; K=0,1,2,3,4** **=0.4 ;K=5** **=0 ;K=6,7** Refer notes		
13	**Explain the procedure of designing FIR filters by window <u>April/May 2018.</u>**Refer notes	**C312.4**	BTL 1
14	**Explain Frequency sampling method of designing FIR filters. <u>April/May 2018</u>** Refer notes	**C312.4**	BTL 1
15	**Given H(Z) H(z)=0.5+0.25 z-1+0.75z-1+z-3+0.75z-4+0.25z-5+0.5z-6)Draw the direct form realization and poly phase Realization .** Refer notes	**C312.4**	BTL 3

UNIT V PART A

Q. No.	Questions	CO	Bloom's Level
1	**Express the fraction 7/8 and – 7/8 in sign magnitude, 2's complement and 1's complement.** **Nov/Dec 2006** **Solution:** 7/8 = 0.875 = $(0.111)_2$ is sign magnitude 1's Complement = $(0.111)_2$ 2's Complement = $(0.111)_2$7/8 = -0.875Sign magnitude: $(1.111)_2$'s Complement = $(1.000)_2$ 2's Complement = $(1.001)_2$	**C312.5**	BTL 2
2	**What are the quantization error due to finite word length register in digital filter. APRIL/MAY2015 MAY/JUNE 2016,** Quantization Error :Input quantization error Coefficient quantization error Product quantization error	**C312.5**	BTL 1
3	**Identify the various factors which degrade the performance of the digital filter implementation when finite word length is used May /June 2007 & April/May 2008 & Nov/Dec 2008** Input quantization error Coefficient quantization error Product quantization	**C312.5**	BTL 3
4	**What is meant by limit cycle oscillation in digital filter? May /June 2007 & Nov/Dec 2007 &April/May 2008,May/June 2013,Nov/Dec 2012.** In recursive system when the input is zero or same non-zero constant value the non linearity due to finite precision arithmetic operation may cause periodic oscillation in the output. Thus the oscillation is called as Limit cycle	**C312.5**	BTL 1
5	**Express the fraction (-7/32) in signed magnitude and 2's complement notations using 6 bits. Nov/Dec 2007 &Nov/Dec 2008** In Signed Magnitude: 1.001110 In 2's complement: 1.110010	**C312.5**	BTL 2
6	**Compare fixed & floating point number representation.**	**C312.5**	BTL 2

S.no	Fixed point number	Floating point number
1.	The position of the binary Point is fixed.	The position of the binary Point is variable.
2.	The resolution is uniform throughout the range	The resolution is variable.

7	**Mention the types quantization errors employed in digital ? April/May 2018** 1. Rounding 2. Truncation	**C312.5**	BTL 1
8	**Define Rounding .** Rounding of a b –bit is accomplished by choosing the rounded result as the b – bit number closed to the original number unrounded.	**C312.5**	BTL 1
9	**What is meant by dead band of the filter ? May/June 2012 MAY/JUNE 2016, How to calculate the deadband of an IIR system? Nov/Dec 2018** In the limit cycle the amplitude of the output are confined to a range of value which is called as dead band of the filter.	**C312.5**	BTL 1
10	**What is fixed point number representation.** The position of the binary Point is fixed**.**	**C312.5**	BTL 1
11	**What is floating point number representation.** The position of the binary Point is variable.	**C312.5**	BTL 1
12	**What are the different quantization methods? Nov/Dec 2006** Truncation Rounding	**C312.5**	BTL 1
13	**Define truncation error for signed magnitude representation and for 2 s complement representation.. April/May 2017 Nov/Dec 2017** Truncation is the process of discarding all bits less significant than least significant bit that is retained.	**C312.5**	BTL 1
14	**What is zero input limit cycle oscillation? (Apr 2004, Nov/Dec 2009 [R2004]) April/May 2017 April/May 2018**	**C312.5**	BTL 1

<table>
<tr><td></td><td>Zero Input Limit Cycles :Zero input limit cycles are usually of lower amplitude in comparison with overflow limit cycles. If the system enters to the limit cycles oscillations, it will continue even after input attains zero range. This equation gives steady state noise power due to quantization.</td><td></td><td></td></tr>
<tr><td>15</td><td>**What is the need for sampling and quantization? (Nov/Dec 2009[R2001])**Sampling is used to convert the Continuous Time signal to Discrete Time signal. Quantization is used to round off the nearest integer value.</td><td>**C312.5**</td><td>BTL 1</td></tr>
<tr><td>16</td><td>**What is steady state noise power at the output of an LTI system due to the quantization at the input to L bits? (Nov 2003 ,Apr 2004 & May/June 2009[R2004])**
The steady state noise power is basically the variance of output
$\sigma_P = \sigma_e^2 .1/2\pi\int |H(\omega)|^2 dw$
Here σ_e^2 is the variance of input error signal
$\sigma_v^2 = 2^{-2L} R_{FS}^2 /48 X ½\pi \int |H (\omega)|^2 dw$</td><td>**C312.5**</td><td>BTL 1</td></tr>
<tr><td>17</td><td>**Compare fixed point and floating point representations.(May/Jun 2006 & May/Jun 2009[R2004])**
<table>
<tr><td>Fixed Point Arithmetic</td><td>Floating Point Arithmetic</td></tr>
<tr><td>It covers only the dynamic range.
Compared to FPA, accuracy is poor
Compared to FPA it is low cost and easy to design
It is preferred for real time operation system
Errors occurs only for multiplication
Processing speed is high
Overflow is rare phenomenon</td><td>It covers a large range of numbers
It attains its higher accuracy
Hardware implementation is difficult to
It is not preferred for realtime operations
Truncation and rounding error occur
multiplication and addition
Processing speed is low
Overflow is a range phenomenon</td></tr>
</table></td><td>**C312.5**</td><td>BTL 2</td></tr>
<tr><td>18</td><td>**Express the fraction (-9/32) in sign magnitude, 2’s complement**</td><td>**C312.5**</td><td>BTL 1</td></tr>
</table>

	notations using 6 bits (Nov 2008) Sign magnitude: 1.01001 2’s complement: 1.10111		
19	**What are the three types of quantization error occurred in digital systems? Nov 2006 & Apr 2008** Input quantization err coefficient quantization error product quantization	**C312.5**	BTL 1
20	**Express the fraction (-7/32) in signed magnitude and two’s complement notations using 6 bits. (Nov 2007)** Sign magnitude: 1.00111 2’s complement: 1.11001	**C312.5**	BTL 1
21	**Express the fraction 7/8 and -7/8 in sign magnitude, 2’s complement and 1’s complement. (Nov 2006)** Sign magnitude: 0.111 1.111 1’s complement: 0.000 1.000 2’s complement: 0.001 1.001	**C312.5**	BTL 2
22	**Define Sampling rate conversion. (May 2007)** Sampling rate conversion is the process of converting a signal from one sampling rate to another, while changing the information carried by the signal as little as possible. Sample rate conversion needed because different systems use different sampling rates**.**	**C312.5**	BTL 1
23	**Convert the number 0.21 into equivalent 6-bit fixed point number. (May 2007)** 0.001101	**C312.5**	BTL 1
24	**Why rounding is preferred to truncation in realizing digital filter?(May2007)**Error introduced due to rounding operation is less compared to truncation. Similarly quantization error due to rounding is independent of arithmetic operation. And mean of rounding error is zero. Hence rounding is preferred over truncation in realizing digital filter.	**C312.5**	BTL 1
25	**What are the different quantization methods? (Nov 2006)** Amplitude quantization , vector quantization , scalar quantization	**C312.5**	BTL 1

26	**What is zero padding? Does zero padding improve the frequency resolution in the spectral estimate? (Nov 2006)** The process of lengthening a sequence by adding zero—valued samples is called appending with zeros or zero padding	**C312.5**	BTL 1
27	**How can overflow limit cycles be eliminated? (Nov 2004)** Saturation Arithmetic , Scaling	**C312.5**	BTL 1
28	**What is meant by finite word length effects in digital filters? (Nov 2003)** The digital implementation of the filter has finite accuracy. When numbers are represented in digital form, errors are introduced due to their finite accuracy. These errors generate finite precision effects or finite word length effects. When multiplication or addition is performed in digital filter, the result is to be represented by finite word length (bits). Therefore the result is quantized so that it can be represented by finite word register. This quantization error can create noise or oscillations in the output. These effects are called finite word length effects.	**C312.5**	BTL 1
29	**What is round – off noise error?** Rounding operation is performed only on magnitude of the number. Hence round-off noise error is independent of type of fixed point representation. If the number is represented by b_u bits before quantization and b bits after quantization, the maximum round-off error will be $(2^{-b} – 2^{-bu})/2$. It is symmetric about zero.	**C312.5**	BTL 1
30	**List the advantages of floating point arithmetic. (Nov2006)** Large dynamic range Occurrence of overflow is very rare Higher accuracy	**C312.5**	BTL 1
31	**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.	**C312.5**	BTL 1
32	**What is transposition theorem & transposed structure?** The transpose of a structure is defined by the following operations. • Reverse the directions of all branches in the signal flow graph • Interchange the input and outputs. • Reverse the roles of all nodes in the flow graph. • Summing points become branching points. • Branching points become summing points.	**C312.5**	BTL 1

	According to transposition theorem if we reverse the directions of all branch transmittance and interchange the input and output in the flowgraph, the system function remains unchanged.		
33	**what are the different types of arithmetic in digital systems.?** There are three types of arithmetic used in digital systems. They are fixed point arithmetic, floating point ,block floating point arithmetic.		BTL 1
34	**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.	**C312.5**	BTL 1
35	**What are the different types of fixed point arithmetic?** Depending on the negative numbers are represented there are three forms of fixed point arithmetic. They are sign magnitude,1's complement,2's complement	**C312.5**	BTL 1
36	**What is meant by sign magnitude representation?** For sign magnitude representation the leading binary digit is used to represent the sign. If it is equal to 1 the number is negative, otherwise it is positive.	**C312.5**	BTL 1
37	**What is meant by 1's complement form?** In 1,s complement form the positive number is represented as in the sign magnitude form. To obtain the negative of the positive number ,complement all the bits of the positive number.	**C312.5**	BTL 1
38	**What is meant by 2's complement form?** In 2's complement form the positive number is represented as in the sign magnitude form. To obtain the negative of the positive number ,complement all the bits of the positive number and add 1 to the LSB. 88. What is meant by floating pint representation? In floating point form the positive number is represented as F =2CM,where is mantissa, is a fraction such that1/2<M<1and C the exponent can be either positive or negative.	**C312.5**	BTL 1
39	**What are the advantages of floating pint representation?** 1.Large dynamic range 2.overflow is unlikely.	**C312.5**	BTL 1
40	**What are the quantization errors due to finite word length registers in digital filters?** 1.Input quantization errors2.Coefficient quantization errors3.Product	**C312.5**	BTL 1

	quantization errors		
41	**What is input quantization error?.** The filter coefficients are computed to infinite precision in theory. But in digital computation the filter coefficients are represented in binary and are stored in registers. If a b bit register is used the filter coefficients must be rounded or truncated to b bits ,which produces an error.	**C312.5**	BTL 1
42	**What is product quantization error?.** The product quantization errors arise at the out put 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 produces the error.	**C312.5**	BTL 1
43	**What is input quantization error?.** The input quantization errors arise due to A/D conversion.	**C312.5**	BTL 1
44	**Distinguish between truncation and Rounding of binary digits with examples. Nov/Dec 2017** Truncation is a process of discarding all bits less significant than LSB that is retained. Rounding a number to b bits is accomplished by choosing a rounded result as the b bit number closest number being unrounded	**C312.5**	BTL 2
45	**Perform the addition of the decimal numbers 0.5 and 0.25 using binary fixed point representation. Nov/Dec 2018**	**C312.5**	BTL 1
46	**What are the two types of limit cycle behavior of DSP?.** 1.Zero limit cycle behavior 2.Over flow limit cycle behavior	**C312.5**	BTL 1
47	**What is truncation?** Truncation is a process of discarding all bits less significant than LSB that is retained.	**C312.5**	BTL 1
48	**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.	**C312.5**	BTL 1
49	**List the disadvantages of floating point arithmetic.** Large dynamic range Occurrence of overflow is very rare Higher accuracy	**C312.5**	BTL 1

50	**What is meant by signal scaling? Nov/Dec2017**To prevent overflow limit cycle oscillation signal scaling is used. Input signal is multiplied by scaling element.	**C312.5**	BTL 1

PART – B

Q. NO.	QUESTIONS	CO	Bloom's Level
1	**Determine the characteristics of a limit cycle oscillation with respect to the system described by the difference equation y(n) =0.95y(n-1)+x(n). Determine the dead band of the filter**. When x(n)+0.875 for n=0 and y(-1)=0 .Assume 4 bit sign magnitude representation.**(Nov2006, May/Jun 2009 [R2004]) April/May 2017 Nov/Dec 2017 April/May 2018 Nov/DEC 2018**Refer Notes	**C312.5**	BTL 5
2	**Draw the quantization noise model for a second order System with system function. (APR 05 EC, Nov/Dec 2009 [R2004]) Nov/Dec2018** $H(z) = 1/[1 - 2r\cos 0\, z^{-1} + r^2\ z^{-2}]$Refer Notes	**C312.5**	BTL 2
3	**Explain the various error introduced due to quantization.(Nov/Dec 2009 [R2001])Nov/Dec 2017 ,April/May 2018** Refer Notes	**C312.5**	BTL 2
4	**Write in detail on Limit Cycle Oscillations. (Nov/Dec 2009 [R2001], Nov/Dec 2009[R2001] April/may 2018)**Refer Notes	**C312.5**	BTL 1
5	**What is the need for signal scaling? How the overflow error scaling is performed? (May/Jun 2009[R2004]) Nov/Dec 2017 Nov/Dec2018.**Refer Notes	**C312.5**	BTL 1
6	**Explain in detail about the zero-input limit cycle oscillations due to finite word length of registers. (May/Jun 2009[R2004])**Refer Notes	**C312.5**	BTL 2
7	**Realize the first order transfer function $H(z) = 1 / (1-az^{-1})$ and draw its quantization model. Find the steady state noise power due to product round off. (May/Jun 2009 [R2004]) MAY/JUNE 2016, How the scaling is performed in Digital filters? Nov/Dec 2017** Refer Notes	**C312.5**	BTL 3

8	**Explain about fixed point and floating point representation**. **(NOV 04 EC, May/Jun 2009[R2001])** Refer booK: Digital Signal Processing by proakis . (pg no6.38&6.39)	**C312.5**	BTL 2
9	**Write notes on quantization noise. Dervie the formula for noise power. (May/Jun** 2009 **.Nov/Dec 2018** [R2001]) Refer booK: Digital signal processing Proakis (pgno: 743)	**C312.5**	BTL 1
10	**(i)Consider (b+1)-bit (including sign bit) bipolar ADC. Obtain an expression for signal to quantization noise ratio. State the assumptions made. (Nov 2008**) Refer booK: Digital signal processing Proakis (pgno: 753)	**C312.5**	BTL 3
11	**Two first order filters are connected in cascaded whose system functions of the individual sections are H1(Z)=1/(1-0.5 Z)AND H1(Z)=1/(1-0.4 Z).Determine the overall output noise power**.**MAY/JUNE 2016. April/May 2017** Refer Notes.	**C312.5**	BTL 4
12	**Explain in detail about finite word length effects in digital filters. April/May 2017 .** Refer Notes	**C312.5**	BTL 2
13	**Bring out the difference between Fixed point and Floating point arithmetic.** **April/May 2017** Refer Notes	**C312.5**	BTL 2
14	**Derive the formula for noise power. How the scaling is performed in Digital filters? Nov/Dec 2017** Refer Notes	**C312.5**	BTL 2
15	**Consider the truncation of negative fraction numbers represented in (β+1) – bit fixed point binary form including sign bit. Let (β-b) bits be truncated. Obtain the range of truncation errors for signed magnitude. 2's complement and 1's complement representations of the negative numbers. (Nov 2007, Nov 2008)**Refer book: Digital signal processing by proakis (Pg no: .21) Refer Notes	**C312.5**	BTL 4