INSTITUTE OF AERONAUTICAL ENGINEERING
(Autonomous)
Dundigal, Hyderabad - 500 043

Department of Electrical and Electronics Engineering

QUESTION BANK

Course Title	DIGITAL SIGNAL PROCESSING
Course Code	A70421
Class	IV B.Tech I Semester
Branch	EEE
Year	2018 - 2019
Course Faculty	Mr. A Naresh Kumar, Assistant Professor, EEE

OBJECTIVES

To meet the challenge of ensuring excellence in engineering education, the issue of quality needs to be addressed, debated and taken forward in a systematic manner. Accreditation is the principal means of quality assurance in higher education. The major emphasis of accreditation process is to measure the outcomes of the program that is being accredited.

In line with this, faculty of Institute of Aeronautical Engineering, Hyderabad has taken a lead in incorporating philosophy of outcome based education in the process of problem solving and career development. So, all students of the institute should understand the depth and approach of course to be taught through this question bank, which will enhance learner's learning process

S. No	QUESTION	Blooms Taxonomy Level	Course Outcome
UNIT - I INTRODUCTION
Part - A (Short Answer Questions)
1	Define symmetric and anti symmetric signals.	Remember	1
2	Explain about impulse response?	Understand	7
3	Understand an LTI system?	Understand	6
4	List the basic steps involved in convolution?	Remember	2
5	Discuss the condition for causality and stability?	Understand	1
6	State the Sampling Theorem	Remember	1
7	Express and sketch the graphical representations of a unit impulse, step	Understand	6
8	Model the Applications of DSP?	Understand	2
9	Develop the relationship between system function and the frequency Response.	Understand	6
10	Discuss the advantages of DSP?	Understand	1
11	Explain about energy and power signals?	Understand	1
12	State the condition for BIBO stable?	Remember	2
13	Define Time invariant system.	Remember	2
14	Define the Parseval's Theorem	Remember	2
15	List out the operations performed on the signals.	Remember	1

16	Discuss about memory and memory less system?	Understand	2
17	Define commutative and associative law of convolutions.	Remember	1
18	Sketch the discrete time signal x(n) =4 δ (n+4) + δ(n)+ 2 δ (n-1) + δ (n-2)	Understand	2
19	Identify the energy and power of $x(n) = Ae^{j\omega n} u(n)$.	Understand	3
20	Illustrate the aliasing effect? How can it be avoided?	Understand	1
21	Define Z-transform and region of converges.	Understand	2
22	Define Z-transform and region of converges.	Understand	4
23	What are the properties of ROC.	Remember	2
24	Write properties of Z-transform.	Understand	2
25	Find z-transform of a impulse and step signals.	Remember	6
26	what are the different methods of evaluating inverse Z-transform	Remember	2
27	Define system function	Understand	2
28	Find The Z-transform of the finite-duration signal x(n)={1,2,5,7,0,1}	Understand	2
29	What is the difference between bilateral and unilateral Z-transform	Remember	2
30	What is the Z-transform of the signal x(n)=Cos(w_0n) u(n).	Remember	2
31	With reference to Z-transform, state the initial and final value theorems?	Remember	2
32	What are the basic building blocks of realization structures?	Understand	4
33	Define canonic and non-canonic structures.	Remember	4
34	Draw the direct-form I realization of 3rd order system	Understand	4
35	What is the main advantage of direct-form II realization when compared to Direct-form I realization?	Remember	4
36	what is advantage of cascade realization	Remember	4
37	Draw the parallel form structure of IIR filter	Understand	4
38	Draw the cascade form structure of IIR filter	Understand	4
39	what is trasposition theorem and transposed structure	Remember	4
40	Transfer function for IIR Filters	Understand	4
41	Transfer function for FIR Filters	Remember	4

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Part - B   (Long Answer Questions)

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1	Identify linear system in the following: a) $y(n) = e_{x(n)}$ b) $y(n) = x_2(n)$ c) $y(n) = ax(n) + b$ d) $y(n) = x(n_2)$	Understand	1
2	Identify a time-variant system. a) $y(n) = e_{x(n)}$ b) $y(n) = x(n_2)$ c) $y(n) = x(n) - x(n-1)$ d) $y(n) = nx(n)$	Understand	2
3	Identify a causal system. a) $y(n) = x(2n)$ b) $y(n) = x(n) - x(n-1)$ c) $y(n) = nx(n)$ d) $y(n) = x(n) + x(n+1)$	Remember	1
4	Determine the impulse response and the unit step response of the systems Understandd by the difference equation y(n) = 0.6y(n-1)-0.08 y(n-2) + x(n).	Understand	2
5	The impulse response of LTI system is h(n)={1 2 1-1} Determine the response of the system if input is x(n)={1 2 3 1}	Remember	1
6	Determine the output y(n) of LTI system with impulse response H (n)= a^n u(n). $\|a\| < 1$ When the input is unit input sequence that is	Remember	1

7	Determine impulse response for cascade of two LTI systems havimg Impulse responses of $H_1(n)=(1/2)^n u(n)$ $H_2(n)=(1/4)^n u(n)$	Understand	1
8	Given the impulse response of a system as $h(k)=a^k u(k)$ determine the range of 'a' for which the system is stable	Remember	2
9	Determine the range of 'a' and 'b'for which the system is stable with impulse response $H(n)= a^n$ $n\geq 0$ b^n $n<0$	Understand	1
10	For each impulse response listed below determine whether the corresponding system is (i) causal (ii) stable a) $h(n)=2nu(-n)$ $\frac{n\pi}{2}$ c)$h(n)=\delta(n)+\sin\pi n$ d) $h(n)=e2nu(n-1)$	Understand	1
11	Find the response of the following difference equation i)$y(n)+y(n-1)=x(n)$ where $x(n)=\cos 2n$ ii)$y(n)-5y(n-1)+6y(n-2)=x(n)$ for $x(n)=n$	Understand	2
12	Find the input $x(n)$ of the system if the impulse response $h(n)$ and output $y(n)$ are shown below $h(n)=\{$ 1 2 3 2$\}$ $y(n)=\{$ 1 3 7 10 10 7 2$\}$	Remember	2
13	Determine the convolution of the pairs of signals by means of z-transform $X1(n)=(1/2)n\ u(n)$ $X2(n)=\cos\pi n\ u(n)$	Remember	2
14	Determine the transfer function and impulse response of the system $y(n)-\ y(n-1)+$ $y(n-2)=x(n)+\ x(n-1)$.	Remember	2
15	Obtain the Direct form II $y(n) = -0.1(n-1) + 0.72\ y(n-2) + 0.7x(n)\ -0.252\ x(n-2)$	Understand	4
16	Find the direct form II $H(z) =8z-2+5z-1+1\ /\ 7z-3+8z-2+1$	Remember	4
Part – C (Analytical Questions)			
1	a) Show that the fundamental period N_p of the signals $s_k(n)= e^{j2\pi kn/N}$ for k=0 2 ….. is given by $N_p = N/GCD(k)$ where GCD is the greatest common divisor of k and N. b) What is the fundamental period of this set for N=7? c) What is it for N=16?	Remember	1
2	Consider the simple signal processing system shown in below figure. The sampling periods of the A/D and D/A converters are T=5ms and T'= 1ms respectively. Determine the output $y_a(t)$ of the system. If the input is $x_a(t) =3 \cos 100\pi t + 2 \sin 250\pi t$ (t in seconds)	Understand	1
3	The post filter removes any frequency component above $F_s/2$. Determine the response $y(n)$	Understand	2
4	Consider the interconnection of LTI systems as shown below. a) Express the overall impulse response in terms of $h_1(n)$ $h_3(n)$ and $h_4(n)$ b) Determine h(n) when $h_1(n)=\{1/2\ 1/2\}$ $h_2(n)=h_3(n)=(n+1)u(n)$ $h_4(n)=\delta(n-2)$ c) Determine the response of above system if $x(n)= \delta(n+2)+3\ \delta(n-1)-4\ \delta(n-3)$	Understand	2
5	Use the one-sided Z-transform to determine y(n) n $\geq$ 0 in the following cases. (a) $y(n)-1.5y(n-1) +0.5y(n-2) = 0$; $y(-1) = 1$; $y(-2) = 0$ (a) Compute the 10 first samples of its impulse response. (b) Find the input-output relation. (c) Understand the input $x(n) = \{1\ 1. ...\}$ and compute the first 10 samples of the output. (d) Compute the first 10 samples of the output for the input given in part (c) by using convolution. (e) Is the system causal? Is it stable?	Understand	2

6	Use the one-sided Z-transform to determine y(n) n ≥ 0 in the following cases. (a) y(n) +y(n−1)−0.25y(n−2) = 0; y(−1) = y(−2) = 1	Understand	2
7	Prove that the fibonacci series can be thought of as the impulse response of the system Understandd by the difference equation y(n) = y(n−1) +y(n−2) +x(n) Then determine h(n) using Z-transform techniques	Remember	2
8	Obtain the i) Direct forms ii) cascade iii) parallel form realizations for the following systems y (n) = 3/4(n-1) − 1/8 y(n-2) + x(n) +1/3 x(n-1)	Understand	4
9	Find the direct form –I cascade and parallel form for H(Z) = z -1 -1 / 1 − 0.5 z-1+0.06 z-2	Remember	4
10	For the LTI system Understandd by the flow graph in figure determine the difference equation relating the input x(n) to the output y(n)	Understand	2
11	Sketch the discrete time signal x(n) =4 δ (n+4) + δ(n)+ 2 δ (n-1) + δ (n-2)	Understand	2
12	Identify the energy and power of $x(n) = Ae^{j\omega n} u(n)$.	Understand	1
13	What is the Z-transform of the signal x(n)=Cos(w$_o$n) u(n).	Remember	2
14	Find The Z-transform of the finite-duration signal x(n)={1,2,5,7,0,1}	Understand	2
15	Sketch the discrete time signal x(n) =5 δ (n+6) + δ(n)+ 3 δ (n-1) + δ (n-8)	Understand	2

<table>
<tr><td colspan="4" align="center">UNIT - II
DISCRETE FORUIER SERIES</td></tr>
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<div align="center">Part - A   (Short Answer Questions)</div>

1	Define discrete fourier series?	Remember	8
2	Distinguish DFT and DTFT?	Understand	8
3	Define N-pint DFT of a sequence x(n)?	Remember	8
4	Define N-pint IDFT of a sequence x(n)?	Remember	8
5	State and prove time shifting property of DFT.	Remember	8
6	Examine the relation between DFT & Z-transform.	Remember	8
7	Outline the DFT X(k) of a sequence x(n) is imaginary.	Understand	8
8	Outline the DFT X(k) of a sequence x(n) is real.	Understand	8
9	Explain the zero padding ?what are its uses.	Understand	8
10	Remember about periodic convolution.	Remember	8
11	Define circular convolution.	Remember	8
12	Distinguish between linear and circular convolution of two sequences	Understand	8
13	Demonstrate the overlap-save method	Understand	8
14	Illustrate the sectioned convolution	Understand	8
15	Demonstrate the overlap-add method	Understand	8
16	State the difference between i)overlap-save ii)overlap-add method	Remember	8
17	Compute the values of WNk , When N=8, k=2 and also for k=3.	Understand	2
18	Discuss about power density spectrum of the periodic signal	Understand	3
19	Compute the DTFT of the sequence x(n)=a n u(n) for a<1	Understand	2
20	Show the circular convolution is obtained using concentric circle method?	Understand	3
21	Why FFT is needed?	Remember	11
22	What is the speed improvement factor in calculation 64-point DFT of sequence using direct computation and FFT algorithm?	Understand	11
23	What are the main advantages of FFT?	Understand	11
24	Determine N=2048, the number of multiplications required using DFT is	Remember	11

25	Determine N=2048, the number of multiplications required using FFT is.	Remember	11
26	Determine, the number of additions required using DFT is.	Remember	11
27	Determine N=2048, the number of additions required using FFT is.	Remember	11
28	What is FFT?	Remember	11
29	What is radix-2 FFT?	Remember	11
30	What is decimation –in-time algorithm?	Remember	11
31	What is decimation –in frequency algorithm?	Remember	11
32	What are the differences and similarities between DIF and DIT algorithms?	Remember	11
33	What is the basic operation of DIT algorithm?	Remember	11
34	What is the basic operation of DIF algorithm?	Remember	11
35	Draw the butterfly diagram of DIT algorithm?	Remember	11
36	How can we calculate IDFT using FFT algorithm?	Understand	11
37	Draw the 4-point radix-2 DIT-FFT butterfly structure for DFT?	Remember	11
38	Draw the 4-point radix-2 DIF-FFT butterfly structure for DFT?	Understand	11
39	What are the Understands of FFT algorithms?	Remember	11
40	Draw the Radix-N FFT diagram for N=6?	Understand	11
Part - B (Long Answer Questions)			
1	Determine the fourier series spectrum of signals i) x(n)=cos$\sqrt{2}\pi$n ii) cosπn/3 iii) x(n) is periodic with period N=4 and x(n)={1 1 0 0}	Remember	8
2	Determine fourier transform and sketch energy density spectrum of signal X(n)=|a| -1<a<1	Remember	8
3	Determine fourier transform and sketch energy density spectrum of signal X(n)= A 0≤n≤L-10 otherwise	Remember	8
4	Derive relation between fourier transform and z-transform	Remember	8
5	Let X(k) be a 14-point DFT of a length 14 real sequence x(n).The first 8 samples of X(k) are given by X(0)=12 X(1)=-1+j3 X(2)=3+j4 X(3)=1-j5 X(4)=-2+2j X(5)=6+j3 X(6)=-2-j3 X(7)=10.Determine the remaining samples	Understand	8
6	Compute DFT of a sequence $(-1)^n$ for N=4	Understand	8
7	Find the DFT of casual 3-sample averager	Understand	8
8	Find the DFT of non-casual 3-sample averager	Understand	8
9	Find 4-point DFT of the following sequences (a) x(n)={1 -1 0 0} (b) x(n)={1 1 -2 -2} (c) x(n)=2^n (d) x(n)=sin(nΠ/2)	Remember	8
10	Determine the circular convolution of the two sequences x1(n)={1 2 3 4} x2(n)={1 1 1 1} and prove that it is equal to the linear convolution of the same	Understand	8
11	Find the output y(n) of a filter whose impulse response is h(n) = {1 1 1} and input signal x(n) = {3 -1 0 1 3 2 0 1 2 1}. Using Overlap add overlap save method	Understand	8
12	Find the output y(n) of a filter whose impulse response is h(n) = {1 11} and input signal x(n) = {3 -1 0 1 3 2 0 1 2 1}. Using Overlap add method.	Understand	8
13	Determine the impulse response for the cascade of two LTI systems having impulse responses h1(n)=(1/2)^n* u(n) h2(n)=(1/4)^n*u(n)	Understand	8
14	Find the output sequence y(n)if h(n)={1 1 1 1} and x(n)={1 2 3 1} using circular convolution	Understand	8
15	Find the convolution sum of x(n) =1 n = -2 0 1 = 2 n= -1 = 0 elsewhere and h(n) = δ (n) – δ (n-1) + δ(n-2) - δ (n-3)	Remember	8

16	Find the DFT of a sequence x(n)={1 2 3 4 4 3 2 1} using DFT algorithm.	Understand	6
17	Find the 8-pont DFT of sequence x(n)={1 1 1 1 1 0 0 0}	Remember	6
18	Compute the eight-point DFT of the sequence X(n)=10≤n≤70 otherwise by using DIT DIF algorithms	Remember	6
19	Compute 4-point DFT of a sequence x(n)={0 1 2 3} using DIT DIF algorithms	Remember	6
20	Compute IDFT of sequence X(K)={7 -.707-j.707 –j 0.707-j0.707 1 0.707+j0.707 j - .707+j.707}	Remember	6
21	Compute the eight-point DFT of the sequence x(n)={0.5 0.5 0.5 0.5 0 0 0 0} using Radix DIT algorithm	Understand	6
22	Compute the eight-point DFT of the sequence x(n)={0.5 0.5 0.5 0.5 0 0 0 0} using radix DIF algorithm	Understand	6
23	Compute the DFT of a sequence x(n)={1 -1 1 -1} using DIT algorithm	Understand	6
24	Remember and compare the 8-point for the following sequences using DIT-FFT algorithm. a)x1(n)= 1 for -3≤n≤3 b) x2(n)= 1 for 0≤n≤6 0 otherwise 0 otherwise	Understand	11

<p style="text-align:center;"><strong style="color:red;">Part – C (Analytical Questions)</p>

1	The linear convolution of length-50 sequence with a length 800 sequence is to be computed using 64 point DFT and IDFT a) What is the smallest number of DFT and IDFT needed to compute the linear convolution using overlap-add method b) What is the smallest number of DFT and IDFT needed to compute the c) Linear convolution using overlap-save method	Understand	8
2	The DTFT of a real signal x(n) is X(F). How is the DTFT of the following signals related to X(F). (a) y(n)=x(-n) (b) r(n)=x(n/4) (c) h(n) =j^nx(n)	Remember	8
3	Consider the sequences x1(n) = {0 1 2 3 4} x2(n) = {0 1 0 0 0} x3(n) = {1 00 0 0} and their 5 point DFT. (a) Determine a sequence y(n) so that Y(k) =X1(k) X2(k) Is there a sequence x3(n) such that S(k) =X1(k) X3(k)	Remember	8
4	Consider a finite duration sequence x(n) = {0 1 2 3 4} (a) Sketch the sequence s(n) with six-point DFT S(k) = w_2^k X(k) k=0 1 .....6 (b) Sketch the sequence y(n) with six-point DFT Y(k) = Re |X(k)| (c) Sketch the sequence v(n) with six-point DFT V(k) = Im|X(k)|	Remember	8
5	Two eight point sequence x1(n) and x2(n) shown in the Figure below. Their DFTs X1[k] and X2[k]. Find the relationship between them.	Understand	8
6	Find the IDFT of sequence X(k)={4 1-j2.414 0 1-j.414 0 1+j.414 0 1+j2.414} using DIF algorithm	Remember	11
7	Show that the product of two complex numbers (a+jb) and (c+jd) can be performed with three real multiplications and five additions using the Algorithm $x_R =$ (a-b)d+(c-d)a x_I=(a-b)d+(c+d)b where $x=x_R+jx_I = (a+jb)(c+jd)$	Remember	11
8	Explain how the DFT can be used to compute N equi-spaced samples of the z- transform of an N-point sequence on a circle of radius r.	Remember	11
9	Develop a radix-3 decimation-in-time FFT algorithm for N=3^n and draw the corresponding flow graph for N=9. What is the number of required complex multiplications? Can the operations be performed in place?	Understand	11
10	Find the IDFT of sequence X(k)={1 1+j 2 1-2j 0 1+2j 0 1-j} using DIF algorithm	Remember	11
11	Find the IDFT of sequence X(k)={8 1+j2 1-j 0 1 0 1+j 1-j2}	Remember	11

12	Draw the signal flow graph for 16-point DFT using a) DIT algorithm b) DIF algorithm	Remember	11
13	Find the IDFT of a sequence x(n)={0 1 2 3 4 5 6 7} using DIT-FFT algorithm.	Understand	11
14	Given x(n)=2n and N=8 find X(k) using DIT-FFT algorithm.	Understand	11
15	Develop a radix-3 decimation-in-frequency FFT algorithm for $N=3^n$ and draw the corresponding flow graph for N=9. What is the number of required complex multiplications? Can the operations be performed in place?	Understand	11

<table>
<tr><td colspan="4" align="center">UNIT - III
IIR DIGTAL FILTERS</td></tr>
<tr><td colspan="4" align="center">Part - A (Short Answer Questions)</td></tr>
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1	Give the magnitude function of butter worth filter. What is the effect of varying order of N on magnitude and phase response?	Understand	10
2	Give any two properties of butter worth low pass filter	Remember	10
3	what are properties of chebyshev filter	Remember	10
4	Give the equation for the cutoff frequency of butter worth filter	Remember	10
5	What is an IIR filter?	Remember	10
6	What is meant by frequency warping? What is the cause of this effect?	Remember	10
7	Distinguish between butter worth and chebyshev filter	Understand	10
8	How can design digital filters from analog filters	Remember	10
9	what is bilinear transformation and properties of bilinear transform	Remember	10
10	what is impulse invariant method of designing IIR filter	Remember	10

11	Distinguish IIR and FIR filters	Remember	10
12	Distinguish analog and digital filters	Remember	10
13	Give the equation for the order N, major, minor axis of an ellipse in case of chebyshev filter?	Understand	10
14	List the Butterworth polynomial for various orders.	Remember	10
15	Write the various frequency transformations in analog domain?	Remember	10
16	What are the advantages of Chebyshev filters over Butterworth filters?	Understand	10
17	What do you understand by backward difference?	Understand	10
18	Write a note on pre warping?	Remember	10
19	What are the specifications of a practical digital filter?	Remember	10
20	Write the expression for the order of Butterworth filter?	Remember	10
21	What is an FIR filter?	Remember	10
22	Write the expression for the order of chebyshev filter?	Remember	10
23	Give the equation for the order of N of butter worth filter?	Remember	10
24	Write short notes on impulse invariant method.	Remember	10
25	Write short notes on bilinear transformation method.	Remember	10

<table>
<tr><td colspan="4" align="center">Part - B (Long Answer Questions)</td></tr>
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1	Given the specification α_p=1dB, α_s=30dB,Ω_p=200rad/sec, Ω_s=600rad/sec. Determine the order of the filter.	Understand	10
2	Determine the order and the poles of lowpass butter worth filter that has a 3 dB attenuation at 500Hz and an attenuation of 40dB at 1000Hz.	Remember	10
3	Design an analog Butterworth filter that as a -2dB pass band attenuation at a frequency of 20rad/sec and at least -10dB stop band attenuation at 30rad/sec.	Understand	10

4	For the given specification design an analog Butterworth filter $.9 \le \mid H(j\Omega) \mid \le 1$ for $0 \le \Omega \le 0.2\pi$ $\mid H(j\Omega) \mid \le 0.2\pi$ for $0.4\pi \le \Omega \le \pi$	Remember	10	
5	For the given specifications find the order of butter worth filter α_p=3dB, α_s=18dB, f_p=1KHz, f_s=2KHz.	Remember	10	
6	Design an analog butter worth filter that has α_p=0.5dB, α_s=22dB, f_p=10KHz, f_s=25KHz Find the pole location of a 6th order butter worth filter with Ω_c=1 rad/sec	Understand	10	
7	Given the specification α_p=3dB, α_s=16dB, f_p=1KHz, f_s=2KHz. Determine the order of the filter Using chebyhev approximation. find H(s).	Understand	10	
8	Obtain an analog chebyshev filter transfer function that satisfies the constraints $0 \le \mid H(j\Omega) \mid \le 1$ for $0 \le \Omega \le 2$	Remember	10	
9	Determine the order and the poles of type 1 low pass chebyshev filter that has a 1 dB ripple in the pass band and pass band frequency Ω_p =1000π and a stop band of frequency of 2000π and an attenuation of 40dB or more.	Understand	10	
10	For the given specifications find the order of chebyshev-I α_p=1.5dB, α_s=10dB, Ω_p=2rad/sec, Ω_s =30 rad/sec.	Remember	10	
11	For the analog transfer function H(s)= Determine H(z) using impulse invariance method .Assume T=1sec $\dfrac{2}{(s+1)(s+2)}$	Understand	10	
12	For the analog transfer function H(s)= $\dfrac{1}{s^2 + \sqrt{2}s + 1}$ Determine H(z) using impulse invariance method .Assume T=1sec	Remember	10	
13	Design a third order butter worth digital filter using impulse invariant technique .Assume sampling period T=1sec	Understand	10	
14	An analog filter has a transfer function H(s)= $\dfrac{2}{(s+1)(s+2)}$ Design a digital filter equivalent to this using impulse invariant method for T=1Sec	Remember	10	

Part – C (Analytical Questions)

1	Given the specification α_p=1dB, α_s=30dB,Ω_p=200rad/sec, Ω_s=600rad/sec. Determine the order of the filter	Understand	10	
2	Determine the order and the poles of low pass butter worth filter that has a 3 dB attenuation at 500Hz and an attenuation of 40dB at 1000Hz	Remember	10	
3	Design an analog Butterworth filter that as a -2dB pass band attenuation at a frequency of 20rad/sec and at least -10dB stop band attenuation at 30rad/sec	Understand	10	
4	For the given specification design an analog Butterworth filter $.9 \le \mid H(j\Omega) \mid \le 1$ for $0 \le \Omega \le 0.2\pi$ $\mid H(j\Omega) \mid \le 0.2\pi$ for $0.4\pi \le \Omega \le \pi$	Remember	10	
5	For the given specifications find the order of butter worth filter α_p=3dB, α_s=18dB, f_p=1KHz, f_s=2KHz.	Remember	10	
6	Design an analog butter worth filter that has α_p=0.5dB, α_s=22dB, f_p=10KHz, f_s=25KHz Find the pole location of a 6th order butter worth filter with Ω_c=1 rad/sec	Understand	10	
7	Given the specification α_p=3dB, α_s=16dB, f_p=1KHz, f_s=2KHz. Determine the order of the filter Using chebyshev approximation. find H(s).	Understand	10	
8	Obtain an analog chebyshev filter transfer function that satisfies the constraints $0 \le \mid H(j\Omega) \mid \le 1$ for $0 \le \Omega \le 2$	Remember	10	
9	Determine the order and the poles of type 1 low pass chebyshev filter that has a 1 dB ripple in the pass band and pass band frequency Ω_p =1000π and a stop band of frequency of 2000π and an attenuation of 40dB or more.	Remember	10	

10	For the given specifications find the order of chebyshev-I α_p=1.5dB, α_s=10dB, Ω_p =2rad/sec, Ω_s =30 rad/sec.	Remember	10
11	For the analog transfer function H(s)= $\dfrac{2}{(s+1)(s+2)}$ Determine H(z) using impulse invariance method .Assume T=1sec	Understand	10
12	For the analog transfer function H(s)= $\dfrac{1}{s^2+\sqrt{2}s+1}$ Determine H(z) using impulse invariance method .Assume T=1sec	Remember	10
13	An analog filter has a transfer function H(s)= $\dfrac{2}{(s+1)(s+2)}$ Design a digital filter equivalent to this using impulse invariant method for T=1Sec	Remember	10
14	For the analog transfer function H(s)= $\dfrac{2}{(s+1)(s+2)}$ using bilinear method Assume T=1sec	Remember	10
15	For the analog transfer function H(s)= $\dfrac{2}{(s+1)(s+2)}$ Determine H(z) using bilinear method Assume T=1sec	Remember	10

UNIT - IV
FIR DIGTAL FILTERS

Part - A (Short Answer Questions)

1	What is mean by FIR filter? and What are advantages of FIR filter?	Understand	13
2	What is the necessary and sufficient condition for the linear phase characteristic of a FIR filter?	Remember	13
3	List the well known design technique for linear phase FIR filter design?	Understand	13
4	For what kind of Understand, the symmetrical impulse response can be used?	Remember	13
5	Under what conditions a finite duration sequence h(n) will yield constant group delay in its frequency response characteristics and not the phase delay?	Remember	13
6	What is Gibbs phenomenon?	Understand	13
7	What are the desirable characteristics of the windows?	Understand	13
8	Compare Hamming window with Kaiser window.	Remember	13
9	Draw impulse response of an ideal low pass filter.	Remember	13
10	What is the principle of designing FIR filter using frequency sampling method?	Remember	13
11	For what type of filters frequency sampling method is suitable?	Understand	13
12	What is the effect of truncating an infinite Fourier series into a finite series	Remember	13
13	What is a Kaiser window? In what way is it superior to other window functions?	Understand	13
14	Explain the procedure for designing FIR filters using windows.	Remember	13
15	What are the disadvantage of Fourier series method ?	Remember	13
16	Draw the frequency response of N point Bartlett window	Understand	13
17	Draw the frequency response of N point Blackman window	Understand	13
18	Draw the frequency response of N point Hanning window	Remember	13
19	What is the necessary and sufficient condition for linear phase characteristics in FIR filter.	Remember	13
20	Give the equation specifying Kaiser window.	Remember	13

	Part - B (Long Answer Questions)		
1	Determine the frequency response of FIR filter defined by y(n)=0.25x(n)+x(n-1)+.25x(n-2) Calculate the phase delay and group delay.	Understand	13
2	The frequency response of Linear phase FIR filter is given by $H(e^{jw})=\cos(w/2+1/2)+\cos 3w/2$. Determine the impulse response(n).	Remember	13
3	If the frequency response of a linear phase FIR filter is given by $H(e^{jw})= e^{-jw2}(.30+0.5\cos\omega+0.3\cos 2\omega)$ Determine filter coefficients.	Understand	13
4	Design an ideal highpass filter with a frequency respose $H_d(e^{jw})$=1 for $\pi/4\le \mid\omega\mid\le\pi$ 0 for $\mid\omega\mid\le\pi/4$ Find the values of h(n) for N=11.Find H(z).plot magnitude response.	Remember	13
5	Design an ideal bandpass filter with a frequency respose $H_d(e^{jw})$=1 for $\pi/4\le\mid\omega\mid\le 3\pi/4$ 0 for $\mid\omega\mid\le\pi/4$ Find the values of h(n) for N=11.Find H(z).plot magnitude response.	Remember	13
6	Design an ideal band reject filter with a frequency respose $H_d(e^{jw})$=1 for $\mid\omega\mid\le$ and $\mid\omega\mid\ge$ 0 for otherwise Find the values of h(n) for N=11.Find H(z).plot magnitude response.	Understand	13
7	Design an ideal differentiate $H(e^{jw})$=j ω -$\pi\le\omega\le\pi$ Using a) rectangular window b)Hamming window with N=8.plot frequency response in both cases.	Understand	13

8	Determine the filter coefficients h(n) obtained by sampling
	
	        $H_d(ejw)=_e$-j(N-1)ω/2 $0\le\mid\omega\mid\le\pi/2$
	
	        =0 $\pi/2<\mid\omega\mid\le\pi$ for N=7
		Remember	13

9	using frequency sampling method design a bandpass filter with following specifications Sampling frequency F=8000Hz Cut off frequency f_{c1}=1000Hz f_{c2}=3000Hz Determine the filter coefficients for N=7	Remember	13
10	Compare IIR and FIR filters	Remember	13
11	Design an FIR filter approximating the ideal frequency response $Hd(ejw)=e$-j$\omega\alpha$ $\mid\omega\mid\le\pi/6$ =0 $\pi/6\le\mid\omega\mid\le\pi$ for N=13Determine filter coefficients.	Understand	13
12	Using a rectangular window technique design a low pass filter with pass band gain of unity, cutoff frequency of 100Hz and working at a sampling frequency of 5KHz.The length of the impulse response should be7.	Remember	13
13	a) Prove that an FIR filter has linear phase if the unit sample response satisfies the condition h(n)= $\pm$ h(M-1-n), n =0,1,….. M-Also discusssym metric and anti symmetric cases of FIR filter. b) Explain the need for the use of window sequence in the design of FIR filter. Understand the window sequence generally used and compare the properties.	Understand	13
14	Design a HPF of length 7 with cut off frequency of 2 rad/sec using Hamming window. Plot the magnitude and phase response.	Remember	13
15	Explain the principle and procedure for designing FIR filter using rectangular window	Remember	13
	Part – C (Analytical Questions)		
1	Design a filter with Hd (ej$\acute{\omega}$) = e- 3 j$\acute{\omega}$, $\pi/4\le\omega\le\pi/4$ 0 for $\pi/4\le\omega\le\pi$ using a Hamming window with N=7.	Understand	13
2	H (w) =1 for $\mid\omega\mid\le\pi/3$ and $\mid\omega\mid\ge 2\pi/3$ otherwise for N=11. and find the response	Remember	13

3	Design a FIR filter whose frequency response H (e jώ) = 1 π/4 ≤ ω ≤ 3π/4 0 | ω | ≤3 π/4. Calculate the value of h(n) for N=11 and hence find H(z).	Understand	13
4	Design an ideal differentiator with frequency response H (e jώ) = jw -π ≤ ω ≤ π using hamming window for N=8 and find the frequency response.	Remember	13
5	Design an ideal Hilbert transformer having frequency response H (e jώ) = j -π ≤ ω ≤ 0 -j 0 ≤ ω ≤ π for N=11 using rectangular window	Remember	13

<table>
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MULTIRATE DIGITAL SIGNAL PROCESSING</td></tr>
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<tr><td>1</td><td>What is decimation by factor D?</td><td>Understand</td><td>12</td></tr>
<tr><td>2</td><td>What is interpolation by factor I?</td><td>Remember</td><td>12</td></tr>
<tr><td>3</td><td>Find the spectrum of exponential signal?</td><td>Understand</td><td>12</td></tr>
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<tr><td>5</td><td>Find the spectrum of exponential signal interpolated by factor 2</td><td>Remember</td><td>12</td></tr>
<tr><td>6</td><td>Explain the term up sampling and down sampling?</td><td>Understand</td><td>12</td></tr>
<tr><td>7</td><td>What are the Understands of multi rate DSP?</td><td>Understand</td><td>12</td></tr>
<tr><td>8</td><td>What does multirate mean?</td><td>Remember</td><td>12</td></tr>
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<tr><td>11</td><td>What are "decimation" and "down sampling"?</td><td>Understand</td><td>12</td></tr>
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<tr><td>15</td><td>Which signals can be down sampled?</td><td>Remember</td><td>12</td></tr>
<tr><td>16</td><td>What happens if I violate the Nyquist criteria in down sampling or ecimating?</td><td>Understand</td><td>12</td></tr>
<tr><td>17</td><td>Can I decimate in multiple stages?</td><td>Understand</td><td>12</td></tr>
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<tr><td>19</td><td>What computational savings do I gain by using a FIR decimator?</td><td>Remember</td><td>12</td></tr>
<tr><td>20</td><td>How much memory savings do I gain by using a FIR decimator?</td><td>Remember</td><td>12</td></tr>
<tr><td>21</td><td>What are the effects of finite word length in digital filters?</td><td>Remember</td><td>7</td></tr>
<tr><td>22</td><td>List the errors which arise due to quantization process.</td><td>Understand</td><td>9</td></tr>
<tr><td>23</td><td>Discuss the truncation error in quantization process.</td><td>Understand</td><td>9</td></tr>
<tr><td>24</td><td>Write expression for variance of round-off quantization noise?</td><td>Remember</td><td>9</td></tr>
<tr><td>25</td><td>Define limit cycle Oscillations, and list out the types.</td><td>Remember</td><td>9</td></tr>
<tr><td>26</td><td>When zero limit cycle oscillation and Over flow limit cycle oscillation has occur?</td><td>Remember</td><td>9</td></tr>
<tr><td>27</td><td>Why? Scaling is important in Finite word length effect?</td><td>Understand</td><td>12</td></tr>
<tr><td>28</td><td>What are the differences between Fixed and Binary floating point number representation?</td><td>Remember</td><td>12</td></tr>
<tr><td>29</td><td>What is the error range for Truncation and round-off process?</td><td>Remember</td><td>9</td></tr>
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<tr><td>33</td><td>How the multiplication & addition are carried out in floating point arithmetic?</td><td>Understand</td><td>12</td></tr>
<tr><td>34</td><td>What do you understand by input quantization error?</td><td>Remember</td><td>9</td></tr>
</table>

35	What is the relationship between truncation error e and the bits b for representing a decimal into binary?	Remember	9
36	What is meant rounding? Discuss its effect on all types of number representation?	Remember	9
37	What is meant by A/D conversion noise?	Understand	9
38	What is the effect of quantization on pole location?	Remember	9
39	What is meant by quantization step size?	Remember	9
40	How would you relate the steady-state noise power due to quantization and the b bits representing the binary sequence?	Remember	9

Part - B (Long Answer Questions)

1	Derive the expression for decimation by factor D	Understand	9
2	Derive the expression for interpolation by factor I	Remember	9
3	Write notes on sampling rate conversion by a rational factor I/D	Remember	9
4	Write notes on filter design and implementation for sampling rate conversion	Remember	9
5	Explain poly phase filter structures	Remember	9
6	Explain time variant filter structures	Remember	9
7	Write notes on the Understand of multi rate digital signal processing	Remember	9
8	Explain the output noise due to A/D conversion of the input x (n).	Remember	9
9	Write short note on (a) Truncation and rounding (b) Coefficient Quantization.	Remember	9
10	Explain the errors introduced by quantization with necessary expression	Understand	9
11	Discuss the various common methods of quantization. Explain the finite word length effects in FIR digital filters.	Understand	9
12	(i). what is quantization of analog signals? Derive the expression for the quantization error. (ii). Explain coefficient quantization in IIR filter.	Remember	12
13	How to prevent limit cycle oscillations? Explain. what is meant by signal scaling? Explain.	Remember	12
14	Discuss in detail the errors resulting from rounding and truncation.	Remember	9
15	Explain the limit cycle oscillations due to product round off and overflow	Remember	12

Part – C (Analytical Questions)

1	a) Understand the decimation process with a neat block diagram. b) Consider a signal x(n)=sin($\prod$n)U(n). Obtain a signal with an interpolation factor of '2'	Understand	12
2	a) Why multirate digital signal processing is needed? b) Design a two state decimator for the following specifications. Decimation factor = 50 Pass band = 0<f<50 Transitive band = 50≤f≤ 55 Input sampling = 10 KHz Ripple = δ1=0.1,δ2=0.001.	Remember	12
3	a) What are the advantages and drawbacks of multirate digital signal processing b) Design a decimator with the following specification D = 5, δp=,0.025 δs=0.0035, ωs= 0.2$\prod$ Assume any other required data.	Remember	12
4	Design one-stage and two-stage interpolators to meet the following specification: $l=20$ Input sampling rate: 10K Hz Passband: $0 \le F \le 90$ Transition band: $90 \le F \le 100$ Ripple: $\delta_1 = 10^{-2}, \delta_2 = 10^{-3}$	Remember	12

5	Design a linear pahse FIR filter that satisfies the following specifications based on a single- stage and two-stage multirate structure. Input sampling rate: 10K Hz Passband: $0 \le F \le 60$ Transition band: $60 \le F \le 65$ Ripple: $\delta1 = 10\text{-}1, \delta2 = 10\text{-}3$	Remember	12
	FINITE WORDLENGTH EFFECTS		
6	The output of an A/D is fed through a digital system whose system function is H $(z)=1/(1\text{-}0.8z^{\text{-}1})$. Find the output noise power of the digital system.	Remember	12
7	The output of an A/D is fed through a digital system whose system function is $H(Z)=0.6z/z\text{-}0.6$. Find the output noise power of the digital system=8 bits	Remember	12
8	Discuss in detail about quantization effect in ADC of signals. Derive the expression for Pe(n) and SNR.	Understand	12
9	A digital system is characterized by the difference equation $y(n)=0.95y(n\text{-}1)+x(n)$. determine the dead band of the system when x(n)=0 and y(-1)=13.	Understand	12
10	Two first order filters are connected in cascaded whose system functions of the individual sections are $H1(z)=1/(1\text{-}0.8z^{\text{-}1})$ and $H2(z)=1/(1\text{-}0.9z^1)$. Determine the overall output noise power.	Remember	12
11	What are the Understands of multirate digital signal processing. b) Design a interpolator which meet the following specifications. Interpolation factor = 20 Pass band : 0≤f< 90 Transitions band : 90<f< 100 Input sampling rate : 10 KHz , Ripple = $\delta1=0.01, \delta2=0.001$.	Remember	12
12	Explain the characteristics of a limit cycle oscillation with respect to the system Understandd by the equation $y(n) = 0.45y(n-1) + x(n)$ when the product is quantized to 5 – bits by rounding. The system is excited by an input x(n) = 0.75 for n = 0 and x(n) = 0 for n ≠ 0. Also determine the dead band of the filter.	Remember	12
13	Consider the LTI system governed by the equation, $y(n) + 0.8301y(n-1) + 0.7348y(n-2) = x(n-2)$. Discuss the effect of co-efficient quantization on pole location , when the coefficients are quantized by 3-bits by truncation 4-bits by truncation	Remember	12
14	. Derive the signal to quantization noise ratio of A/D converter. (ii). Compare the truncation and rounding errors using fixed point and floating point representation.	Remember	9
15	Understand the quantization in floating point realization of IIR digital filters. (i). Explain the characteristics of limit cycle oscillation with respect to the system Understandd by the difference equation: Y (n) = 0.95y(n-1) + x(n); x(n) = 0 and y(-1) = 13. Determine the dead band range of the system. (ii). Explain the effects of coefficient quantization in FIR filters	Understand	9

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