# INSTITUTE OF AERONAUTICAL ENGINEERING

**(Autonomous)**

**Dundigal, Hyderabad - 500 043**

## Department of Electrical and Electronics Engineering

## QUESTION BANK

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| **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 |
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| **UNIT - I<br>INTRODUCTION** | | | |
| **Part - A (Short Answer Questions)** | | | |
| 1 | Define symmetric and anti symmetric signals. | Remember | 1 |
| 2 | Explain about impulse response? | Understand | 7 |
| 3 | Describe 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? | Describe | 2 |
| 9 | Develop the relationship between system function and the frequency Response. | Describe | 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 |

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| 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) | Describe | 2 |
| 19 | Identify the energy and power of $x(n) = Ae^{j\omega n} u(n)$. | Describe | 3 |
| 20 | Illustrate the aliasing effect? How can it be avoided? | Describe | 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 R O C. | 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 $x(n)=Cos(w_0 n)\ u(n)$. | 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 transposition theorem and transposed structure | Remember | 4 |
| 40 | Transfer function for IIR Filters | Understand | 4 |
| 41 | Transfer function for FIR Filters | Remember | 4 |
| | **Part - B (Long Answer Questions)** | | |
| 1 | Identify linear system in the following:<br>a) $y(n)=e_{x(n)}$ b) $y(n)=x_2(n)$<br>c) $y(n)=ax(n)+b$ d) $y(n)=x(n_2)$ | Understand | 1 |
| 2 | Identify a time-variant system.<br>a) $y(n)=e_{x(n)}$ b) $y(n)=x(n_2)$<br>c) $y(n)=x(n)-x(n-1)$ d) $y(n)=nx(n)$ | Describe | 2 |
| 3 | Identify a causal system.<br>a) $y(n)=x(2n)$ b) $y(n)=x(n)-x(n-1)$<br>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 described by the difference equation y(n) = 0.6y(n-1)-0.08 y(n-2) + x(n). | Describe | 2 |
| 5 | The impulse response of LTI system is<br>h(n)={1 2 1-1} Determine the response of the system if input is x(n)={1 2 31} | Remember | 1 |
| 6 | Determine the output y(n) of LTI system with impulse response<br>H (n)= $a^n$ u(n). $\|a\|<1$ When the input is unit input sequence that is | Remember | 1 |

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| 7 | Determine impulse response for cascade of two LTI systems havimg Impulse responses of<br>$H_1(n)=(1/2)^n u(n)$ $H_2(n)=(1/4)^n u(n)$ | Describe | 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<br>H(n)= $a^n$ n≥0<br>$b^n$ n<0 | Describe | 1 |
| 10 | For each impulse response listed below determine whether the corresponding system is (i) causal (ii) stable<br>a) h(n)=2nu(-n) $\frac{n\pi}{2}$ c)h(n)=δ(n)+sinπn<br>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)=cos2n<br>ii)y(n)-5y(n-1)+6y(n-2)=x(n) for x(n)=n | Describe | 2 |
| 12 | Find the input x(n) of the system if the impulse response h(n) and output y(n)<br>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)= cosπn u(n) | Remember | 2 |
| 14 | Determine the transfer function and impulse response of the system<br>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.<br>b) What is the fundamental period of this set for N=7?<br>c) What is it for N=16? | Rememb er | 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 cost 100πt + 2 sin 250πt ( t in seconds) | Describe | 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.<br>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} $h_2(n)=h_3(n)=(n+1)u(n)$ $h_4(n)=\delta(n-2)$<br>c) Determine the response of above system if x(n)= δ(n+2)+3 δ(n-1)-4 δ(n-3) | Describe | 2 |
| 5 | Use the one-sided Z-transform to determine y(n) n ≥ 0 in the following cases.<br>(a) y(n)−1.5y(n−1) +0.5y(n−2) = 0; y(−1) = 1; y(−2) = 0<br>(a) Compute the 10 first samples of its impulse response.<br>(b) Find the input-output relation.<br>(c) Describe the input x(n) = {1 1. . . .} and compute the first 10 samples of the output.<br>(d) Compute the first 10 samples of the output for the input given in part (c) by using convolution.<br>(e) Is the system causal? Is it stable? | Understand | 2 |

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| 6 | Use the one-sided Z-transform to determine y(n) $n \geq 0$ in the following cases.<br>(a) $y(n) + y(n-1) - 0.25y(n-2) = 0$; $y(-1) = y(-2) = 1$ | Describe | 2 |
| 7 | Prove that the fibonacci series can be thought of as the impulse response of the system<br>described 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<br>y (n) = 3/4(n-1) – 1/8 y(n-2) + x(n) +1/3 x(n-1) | Describe | 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 described 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 $x(n) = 4\delta(n+4) + \delta(n) + 2\delta(n-1) + \delta(n-2)$ | Describe | 2 |
| 12 | Identify the energy and power of $x(n) = Ae^{j\omega n} u(n)$. | Describe | 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 $x(n) = 5\delta(n+6) + \delta(n) + 3\delta(n-1) + \delta(n-8)$ | Describe | 2 |

## UNIT - II
## DISCRETE FORUIER SERIES

### Part - A (Short Answer Questions)

| | | | |
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| 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 | Describe | 8 |
| 14 | Illustrate the sectioned convolution | Describe | 8 |
| 15 | Demonstrate the overlap-add method | Describe | 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. | Describe | 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 | Describe | 2 |
| 20 | Show the circular convolution is obtained using concentric circle method? | Describe | 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 |

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| 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? | Describe | 11 |
| 39 | What are the Describes of FFT algorithms? | Remember | 11 |
| 40 | Draw the Radix-N FFT diagram for N=6? | Describe | 11 |
| **Part - B (Long Answer Questions)** | | | |
| 1 | Determine the fourier series spectrum of signals<br>i) x(n)=cos√2πn<br>ii) cosπn/3<br>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 | Describe | 8 |
| 7 | Find the DFT of casual 3-sample averager | Describe | 8 |
| 8 | Find the DFT of non-casual 3-sample averager | Describe | 8 |
| 9 | Find 4-point DFT of the following sequences<br>(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 | Describe | 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. | Describe | 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) | Describe | 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 | Describe | 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 |

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| 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 | Describe | 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 | Describe | 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<br>0 otherwise 0 otherwise | Describe | 11 |
| | **Part – C (Analytical Questions)** | | |
| 1 | The linear convolution of length-50 sequence with a length 800 sequence is to be computed using 64 point DFT and IDFT<br>a) What is the smallest number of DFT and IDFT needed to compute the linear convolution using overlap-add method<br>b) What is the smallest number of DFT and IDFT needed to compute the<br>c) Linear convolution using overlap-save method | Describe | 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) $=j^n x(n)$ | 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.<br>(a) Determine a sequence y(n) so that Y(k) =X1(k) X2(k)<br>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<br>(b) Sketch the sequence y(n) with six-point DFT Y(k) = Re\|X(k)\|<br>(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. | Describe | 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<br>$x_R = (a-b)d+(c-d)a$<br>$x_I=(a-b)d+(c+d)b$<br>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? | Describe | 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 |

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| 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. | Describe | 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? | Describe | 11 |
| | **UNIT - III**<br>**IIR DIGTAL FILTERS** | | |
| | **Part - A (Short Answer Questions)** | | |
| 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 |
| | **Part - B (Long Answer Questions)** | | |
| 1 | Given the specification $\alpha_p$=1dB, $\alpha_s$=30dB,$\Omega_p$=200rad/sec, $\Omega_s$=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 |

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| 4 | For the given specification design an analog Butterworth filter $.9 \le \lvert H(j\Omega) \rvert \le 1$ for $0 \le \Omega \le 0.2\pi$ $\lvert H(j\Omega) \rvert \le 0.2\pi$ for $0.4\pi \le \Omega \le \pi$ | Remember | 10 |
| 5 | For the given specifications find the order of butter worth filter $\alpha_p$=3dB, $\alpha_s$=18dB, $f_p$=1KHz, $f_s$=2KHz. | Remember | 10 |
| 6 | Design an analog butter worth filter that has $\alpha_p$=0.5dB, $\alpha_s$=22dB, $f_p$=10KHz, $f_s$=25KHz Find the pole location of a $6^{th}$ order butter worth filter with $\Omega_c$=1 rad/sec | Understand | 10 |
| 7 | Given the specification $\alpha_p$=3dB, $\alpha_s$=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 $0 \le \lvert H(j\Omega) \rvert \le 1$ 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 $\Omega_p$ =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 $\alpha_p$=1.5dB, $\alpha_s$=10dB, $\Omega_p$ =2rad/sec, $\Omega_s$ =30 rad/sec. | Remember | 10 |
| 11 | For the analog transfer function H(s)= $\frac{2}{(s+1)(s+2)}$ Determine H(z) using impulse invariance method .Assume T=1sec | Understand | 10 |
| 12 | For the analog transfer function H(s)= $\frac{1}{s^2+\sqrt{2}s+1}$ 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)= $\frac{2}{(s+1)(s+2)}$ Design a digital filter equivalent to this using impulse invariant method for T=1Sec | Remember | 10 |
| **Part – C (Analytical Questions)** | | | |
| 1 | Given the specification $\alpha_p$=1dB, $\alpha_s$=30dB,$\Omega_p$=200rad/sec, $\Omega_s$=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 $.9 \le \lvert H(j\Omega) \rvert \le 1$ for $0 \le \Omega \le 0.2\pi$ $\lvert H(j\Omega) \rvert \le 0.2\pi$ for $0.4\pi \le \Omega \le \pi$ | Remember | 10 |
| 5 | For the given specifications find the order of butter worth filter $\alpha_p$=3dB, $\alpha_s$=18dB, $f_p$=1KHz, $f_s$=2KHz. | Remember | 10 |
| 6 | Design an analog butter worth filter that has $\alpha_p$=0.5dB, $\alpha_s$=22dB, $f_p$=10KHz, $f_s$=25KHz Find the pole location of a $6^{th}$ order butter worth filter with $\Omega_c$=1 rad/sec | Understand | 10 |
| 7 | Given the specification $\alpha_p$=3dB, $\alpha_s$=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 $0 \le \lvert H(j\Omega) \rvert \le 1$ 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 $\Omega_p$ =1000π and a stop band of frequency of 2000π and an attenuation of 40dB or more. | Remember | 10 |

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| 10 | For the given specifications find the order of chebyshev-I $\alpha_p$=1.5dB, $\alpha_s$=10dB, $\Omega_p$ =2rad/sec, $\Omega_s$ =30 rad/sec. | Remember | 10 |
| | | | |
| 11 | For the analog transfer function H(s)= $\frac{2}{(s+1)(s+2)}$ Determine H(z) using impulse invariance method .Assume T=1sec | Understand | 10 |
| 12 | For the analog transfer function H(s)= $\frac{1}{s^2+\sqrt{2}s+1}$ Determine H(z) using impulse invariance method .Assume T=1sec | Remember | 10 |
| 13 | An analog filter has a transfer function H(s)= $\frac{2}{(s+1)(s+2)}$ Design a digital filter equivalent to this using impulse invariant method for T=1Sec | Remember | 10 |
| 14 | For the analog transfer function H(s)= $\frac{2}{(s+1)(s+2)}$ using bilinear method Assume T=1sec | Remember | 10 |
| 15 | For the analog transfer function H(s)= $\frac{2}{(s+1)(s+2)}$ Determine H(z) using bilinear method Assume T=1sec | Remember | 10 |
| **UNIT - IV**<br>**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 Describe, 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. | Remem ber | 13 |
| 20 | Give the equation specifying Kaiser window. | Remem | 13 |
| **Part - B (Long Answer Questions)** | | | |

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| 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 $H(e^{jw})=\cos(w/2+1/2)\ +\cos3w/2$ . Determine the impulse response(n). | Remember | 13 |
| 3 | If the frequency response of a linear phase FIR filter is given by $H(e^{jw})= e^{-jw2}(.30+0.5\cos\omega+0.3\cos2\omega)$ Determine filter coefficients. | Understand | 13 |
| 4 | Design an ideal highpass filter with a frequency respose<br>$H_d(e^{jw})=1$ for $\pi/4\leq |\omega| \leq\pi$<br>0 for $|\omega| \leq \pi/4$<br>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<br>$H_d(e^{jw})=1$ for $\pi/4\leq |\omega| \leq 3\pi/4$<br>0 for $|\omega| \leq \pi/4$<br>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<br>$H_d(e^{jw})=1$ for $|\omega| \leq$ and $|\omega| \geq$ 0 for otherwise<br>Find the values of h(n) for N=11.Find H(z).plot magnitude response. | Understand | 13 |
| 7 | Design an ideal differentiate $H(e^{jw})=j\omega\ -\pi\leq\omega\leq\pi$<br>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<br>$H_d(ejw)=e^{-j(N-1)\omega/2}$ $0\leq|\omega| \leq\pi/2$<br>$=0$ $\pi/2<|\omega| \leq\pi$ 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<br>$Hd(ejw)=e^{-j\omega\alpha}$ $|\omega| \leq \pi/6$ $=0$ $\pi/6 \leq|\omega| \leq\pi$ 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.<br>b) Explain the need for the use of window sequence in the design of FIR filter. Describe the window sequence generally used and compare the properties. | 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 (ejώ) = e- 3 jώ, $\pi/4 \leq \omega \leq \pi/4$ 0 for $\pi/4 \leq \omega \leq \pi$ using a Hamming window with N=7. | Understand | 13 |
| 2 | H (w) =1 for $|\omega| \leq \pi/3$ and $|\omega| \geq 2\pi/3$ otherwise for N=11. and find the response | Remember | 13 |

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| 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 |
| **UNIT - V**<br>**MULTIRATE DIGITAL SIGNAL PROCESSING** | | | |
| **Part - A (Short Answer Questions)** | | | |
| 1 | What is decimation by factor D? | Understand | 12 |
| 2 | What is interpolation by factor I? | Remember | 12 |
| 3 | Find the spectrum of exponential signal? | Understand | 12 |
| 4 | Find the spectrum of exponential signal decimated by factor 2. | Remember | 12 |
| 5 | Find the spectrum of exponential signal interpolated by factor 2 | Remember | 12 |
| 6 | Explain the term up sampling and down sampling? | Understand | 12 |
| 7 | What are the Describes of multi rate DSP? | Understand | 12 |
| 8 | What does multirate mean? | Remember | 12 |
| 9 | Why should I do multirate DSP? | Remember | 12 |
| 10 | What are the categories of multirate? | Remember | 12 |
| 11 | What are "decimation" and "down sampling"? | Understand | 12 |
| 12 | What is the "decimation factor"? | Remember | 12 |
| 13 | Why decimate? | Understand | 12 |
| 14 | Is there a restriction on decimation factors I can use? | Remember | 12 |
| 15 | Which signals can be down sampled? | Remember | 12 |
| 16 | What happens if I violate the Nyquist criteria in down sampling or ecimating? | Understand | 12 |
| 17 | Can I decimate in multiple stages? | Understand | 12 |
| 18 | How do I implement decimation? | Remember | 12 |
| 19 | What computational savings do I gain by using a FIR decimator? | Remember | 12 |
| 20 | How much memory savings do I gain by using a FIR decimator? | Remember | 12 |
| 21 | What are the effects of finite word length in digital filters? | Remember | 7 |
| 22 | List the errors which arise due to quantization process. | Understand | 9 |
| 23 | Discuss the truncation error in quantization process. | Understand | 9 |
| 24 | Write expression for variance of round-off quantization noise? | Remember | 9 |
| 25 | Define limit cycle Oscillations, and list out the types. | Remember | 9 |
| 26 | When zero limit cycle oscillation and Over flow limit cycle oscillation has occur? | Remember | 9 |
| 27 | Why? Scaling is important in Finite word length effect? | Understand | 12 |
| 28 | What are the differences between Fixed and Binary floating point number representation? | Remember | 12 |
| 29 | What is the error range for Truncation and round-off process? | Remember | 9 |
| 30 | What do you understand by a fixed-point number? | Remember | 12 |
| 31 | What is meant by block floating point representation? What are its advantages? | Remember | 12 |
| 32 | What are the advantages of floating point arithmetic? | Understand | 12 |
| 33 | How the multiplication & addition are carried out in floating point arithmetic? | Understand | 12 |
| 34 | What do you understand by input quantization error? | Remember | 9 |

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| 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 Describe 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.<br>Explain the finite word length effects in FIR digital filters. | Describe | 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.<br>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) Describe the decimation process with a neat block diagram.<br>b) Consider a signal x(n)=sin(∏n)U(n). Obtain a signal with an interpolation factor of '2' | Understand | 12 |
| 2 | a) Why multirate digital signal processing is needed?<br>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∏ 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<br>Passband: $0 \leq F \leq 90$ Transition band: $90 \leq F \leq 100$<br>Ripple: $\delta_1=10^{-2}$, $\delta_2=10^{-3}$ | Remember | 12 |

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| 5 | Design a linear pahse FIR filter that satisfies the following specifications based on a single- stage and two-stage multirate structure.<br>Input sampling rate: 10K Hz<br>Passband: $0 \leq F \leq 60$<br>Transition band: $60 \leq F \leq 65$<br>Ripple: δ1 =10-1,δ2 = 10-3 | 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/(1-0.8z-¹).Find the output noise power of the digital system. | Rememb er | 12 |
| 7 | The output of an A/D is fed through a digital system whose system function is H(Z)=0.6z/z-0.6. Find the output noise power of the digital system=8 bits | Rememb er | 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-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/(1-0.8z-¹ ) and H2(z)=1/(1-0.9z¹ ).Determine the overall output noise power. | Remember | 12 |
| 11 | What are the Describes 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 = δ1=0.01,δ2=0.001. | Remember | 12 |
| 12 | Explain the characteristics of a limit cycle oscillation with respect to the system described 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.<br>(ii). Compare the truncation and rounding errors using fixed point and floating point representation. | Remember | 9 |
| 15 | Describe the quantization in floating point realization of IIR digital filters.<br>(i). Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation:<br>Y (n) = 0.95y(n – 1) + x(n); x(n) = 0 and y(-1) = 13. Determine the dead band range of the system.<br>(ii). Explain the effects of coefficient quantization in FIR filters | Understand | 9 |

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