**ST. ANNE’S**

**COLLEGE OF ENGINEERING AND TECHNOLOGY**

(Approved by AICTE, New Delhi. Affiliated to Anna University, Chennai)

(An ISO 9001: 2015 Certified Institution)

ANGUCHETTYPALAYAM,PANRUTI – 607 106.

**QUESTION BANK**

**PERIOD:** JULY - NOV 2018 **BATCH**: 2016 – 2020

**BRANCH:** ECE **YEAR/SEM:** III/V

**SUB CODE/NAME:** EC6502- PRINCIPLE OF DIGITAL SIGNAL PROCESSING

**UNIT I DISCRETE FOURIER TRANSFORM**

**PART-A**

1. What is Digital Signal Processing?**[D]**
2. Write down the equations forward and reverse on DFT? **(MAY/JUNE 2010) [D]**
3. Compare the DIT and DIF radix-2 FFT? **(MAY/JUNE 2010) [D]**
4. How many multiplications and additions are required to compute N-point DFT using radix-2 FFT?**(NOV/DEC 2010) (NOV/DEC 2013) [ID]**
5. Obtain the circular convolution for the following sequence x(n)={1,2,1} and

 h(n)={1,-2,2}.**(NOV/DEC 2010) [D]**

1. State the advantages of FFT over DFT’s.**(APR/MAY 2011) [D]**
2. What is meant by bit reversal? **(APR/MAY 2011) [D]**
3. Distinguish between DTFT and DFT? **(NOV/DEC 2011) [D]**
4. What is zero padding? What are its uses? **(NOV/DEC 2011`)[D]**
5. Distinguish between discrete time Fourier transform and discrete Fourier transform?**(MAY/JUNE2012) [D]**
6. Determine the number of multiplications required in the computation of 8-point

 DFT. **(MAY/JUNE2012) [ID]**

1. What is twiddle factor? Or What is phase factor ?**(NOV/DEC 2012) [D]**
2. List the use of FFT in linear filtering? **(NOV/DEC 2012) [D]**
3. Find the 4-point DFT of the sequences x(n)={1,1,-1,-1}**.(MAY/JUNE2013) [D]**
4. What is meant by ‘in-place’ computation? **(MAY/JUNE2013) [ID]**
5. What is zero padding? What is the purpose of it? **(NOV/DEC 2013) [D]**
6. State the difference betweenDTFT and DFT? **(MAY/JUNE2014) (NOV/DEC 2015) [D]**
7. What is bit reversal?**(MAY/JUNE2014) (NOV/DEC 2015) (MAY/JUNE2016) [D]**
8. Compare the number of multiplications required to compute the DFT of a 64-point sequence using direct computation and that using FFT? **(NOV/DEC 2014) [ID]**
9. What is meant by ‘in-place’ in DIT and DIF algorithm? **(NOV/DEC 2014) [ID]**
10. Write the analysis and synthesis equations of DFT? **(APR/MAY 2015) [ID]**
11. Is the DFT of a finite length sequence periodic? If so, state the theorem? **(APR/MAY 2015) [ID]**
12. State the advantages of FFT over DFT? **[D]**
13. Compare radix-2 DIT, DIF-FFT algorithm? **(NOV/DEC 2016) [D]**
14. Test the causality and stability of y (n) = sin x (n). **(NOV/DEC 2016) [ID]**
15. What is the relation between DTFT and DFT? **(APR/MAY 2017) [D]**
16. Compute the DFT of the sequence x(n)={1,-1,1,-1}**(APR/MAY 2017) [D]**
17. What is twiddle factor? **(NOV/DEC 2017) [D]**
18. State and prove periodicity property of DFT? **(NOV/DEC 2017) [D]**
19. Why FFT is needed? **[D]**
20. What is the main advantage of FFT? **[D]**
21. What is FFT? **[D]**
22. What is the difference between DIF and DIT algorithm? **[D]**
23. What is meant by radix-2 FFT? **[D]**
24. List any two properties of DFT. **[D]**

**PART-B**

**DFT AND ITS PROPERTISE**

1. Determine the 8-point DFT of the signal x(n)={1,1,1,1,1,0,0,0} and sketch its magnitude and phase. **[D] [13]**
2. Determine the DFT of the sequence

 x(n)={1/5, for 0 ≤ n ≤ 2

 0, otherwise **(NOV/DEC 2010) [D] [13]**

1. State any six properties of DFT.**(NOV/DEC 2013) [D] [13]**

**CIRCULAR CONVOLUTION**

1. Perform circular convolution of the sequences,x1(n)={1,2,2,1} and

x2(n)={1,2,3,1}**[D] [13]**

1. Perform circular convolution of the following sequences x(n)={1,1,2,1} and h(n)={1,2,2,1} using DFT and IDFT method. **[D] [13]**
2. Determine the circular convolution of the sequences x1(n) = { 1, 2, 3, 1, 1, 2, 3, 1}

 and x2(n) = {4, 3, 2, 2, 2, 2, 3, 4) using DFT and IDFT. **(NOV/DEC 2017) [D] [13]**

1. In an LTI system the input x(n) = {1, 1, 2, 1} and the impulse response

h(n), {1, 2, 3, 4} . Perform the circular convolution using DFT and IDFT. **(APR/MAY 2017) [D] [13]**

**FILTERING METHODS BASED ON DFT**

**DIT-FFT**

1. Explain in detail about the construction of 8-point Decimation in time FFT algorithm.**(MAY/JUNE 2012) [D] [13]**
2. Compute the eight point DFT of the sequence x(n)={0.5,0.5,0.5,0.5,0,0,0,0}using the in place radix-2 DIT algorithm. **(MAY/JUNE-13) [D] [13]**
3. Find the 8 point DFT of the sequence x(n) = { 2, 2, 2, 2, 1, 1, 1, 1} using

Decimation in Time FFT algorithm. **(NOV/DEC 2017) [D] [13]**

1. Derive radix 2 - DIT FFT algorithm and obtain DFT of the sequence

x(n) = {1,2,3,4,4,3,2,1} using DIT algorithm. **(NOV/DEC 2016) [D] [13]**

1. Compute the 8-point DFT of the sequence x(n)={1;0≤n≤7

 0; otherwise

 by using DIT algorithm. **[ID] [13]**

1. Find DFT for {1,1,2,0,1,2,0,1} using FFT DIT butterfly algorithm.**(NOV/DEC-2014) [D] [13]**
2. Using radix -2 DIT-FFT algorithm, determine DFT of the given sequence for N=8

X(n)={n; for n≤7

 0; otherwise**[ID] [13]**

1. Compute the eight point DFT of the sequence x(n)={1,-1,-1,-1,1,1,1,-1}using the in place radix-2 DIT algorithm. **[D] [13]**

**DIF-FFT**

1. Compute the eight point DFT of the sequence x(n)={0.5,0.5,0.5,0.5,0,0,0,0}using the in place radix-2 DIF algorithm. **[D] [13]**
2. Explain in detail about the construction of 8-point Decimation in Frequency FFT algorithm. **[D] [13]**
3. Compute the DFT of the sequence x(n)={1,2,3,4,4,3,2,1}.Using radix-2 DIF-FFT algorithm. **(APR/MAY 2017) [D] [13]**
4. Compute IDFT of the sequence X(K) = { 7,- 0.707,- j0.707, - j,

 0.707 - j0.707, 1, 0.707 + j0.707 j, - 0.707 + j 0.707} using DIF algorithm. **(NOV/DEC 2016) [D] [13]**

1. Explain the radix-2 DIF-FFT algorithm? Compare it with DIT-FFT. **(MAY/JUNE2016)[D] [13]**
2. Find the IDFT of the sequence X(k) )={4,1-j2.414,0,1-j0.414,0,1+j0.414,0,1+j2.414} using DIF algorithm. **(MAY/JUNE-13)** **[D][13]**
3. Compute the eight point DFT of the sequence x(n)=n+1using radix-2 DIF algorithm. **(MAY/JUNE-13) [ID] [13]**

**FFT IN LINEAR FILTERING**

1. An FIR digital filter has the unit impulse response sequence, h(n)={2,2,1}. Determine the output sequence in response to the input sequence x(n)={3,0,-2,0,2,1,0,-2,-1,0} using the over-lap and overlap save convolution method. **[ID] [13]**
2. Explain in detail about over-lap add Method **(APR/MAY2011) (NOV/DEC 2010) [D] [13]**
3. Find the output y(n) of a filter whose impulse response h(n)= {1,1,1} and input signal x(n)= {3,-1,0,1,3,2,0,1,2,1} using overlap save method**.(MAY/JUNE-13) [D] [13]**
4. Perform the linear convolution of finite duration sequences

h(n) = {1, 2} and x(n) = {1,2,-1,2,3,- 2,- 3,-1,1, 1, 2,-1} by overlap save method. **(NOV/DEC 2016) [D] [13]**

1. With appropriate diagrams explain
2. Overlap-add method
3. Overlap-save method**(MAY/JUNE2016)[D] [13]**

**UNIT-2 IIR FILTER DESIGN**

**PART A**

1. Convert the analog filter with system function H(s) into digital IIR filter by

H(s)= 1/(S+0.2)(S+0.6) **(APRIL/MAY-2010) [D]**

1. What are limitation of impulse invariance method of designing digital filter**?(NOV/DEC-2010) [D]**
2. Draw the ideal gain Vs frequency characteristics of i) HPF and ii) BPF. **(NOV/DEC-2010) [D]**
3. What is meant by warping**?(APRIL/MAY-2011) [D]**
4. What are limitation of impulse invariance method? **(APRIL/MAY-2011) [D]**
5. Why is the butterworth response called a maximally flat response? **(MAY/JUNE 2012) [D]**
6. What is frequency warping? **(MAY/JUNE 2012) [D]**
7. Draw the direct form realization of IIR system**.(MAY/JUNE-2013) [D]**
8. Distinguish between the frequency response of chebyshev type-1 filter for N odd and N even. **(MAY/JUNE-2013) [D]**
9. Define bilinear transformation with expression.(**NOV/DEC-2013) [D]**
10. Mention the properties of Buttterworth filter. **.(NOV/DEC-2013) [D]**
11. Mention the properties of Chebyshev filter**. (NOV/DEC-2013) [D]**
12. What is Warping ? What is its effect on magnitude and phase response. **(MAY/JUNE-2014) [D]**
13. What are the properties of impulse invariance transformation?**(MAY/JUNE-14) [D]**
14. What is meant by bilinear transformation method of designing IIR filter? **[D]**
15. Compare analog and digital filters**.(NOV/DEC-2014) [D]**
16. Sketch the mapping of S-plane and Z-plane in approximation of derivatives. **(NOV/DEC-2014) [D]**
17. What is prewarping? .**(NOV/DEC-2014) [D]**
18. Distinguish between butterworth and chebyshev filter**.(NOV/DEC-2014) [D]**
19. Find H(z) for the IIR filter whose H(s)=1/(s+6) with T=0.1 sec. **(APRIL/MAY-2015)**
20. Draw the response curve for butterworth, chebychev and elliptic filters. **(APRIL/MAY-2015) [D]**
21. Distinguish between butterworth and chebyshev filter**.(NOV/DEC-2015) [D]**
22. What is prewarping? **(NOV/DEC-2015) [D]**
23. Mention the advantages of cascade realization? **(MAY/JUNE-2016) [D]**
24. Convert the given analog transfer function H(s) = 1/ (s+a) into digital by impulse invariant method. **(MAY/JUNE-2016) [D]**
25. What is known as prewarping? **(NOV/DEC-2016) [D]**
26. What are the properties of bilinear transformation? **(NOV/DEC-2016) [D]**
27. What are the requirements for the digital filter to be stable and casual? **(APR/MAY 2017) [ID]**
28. Discuss the need for prewarping. **(APR/MAY 2017) [D]**
29. List the different types of filters based on frequency response. **(NOV/DEC-2017) [D]**
30. What are the properties of bilinear transformation ? **(NOV/DEC-2017) [D]**
31. Find digital filter equivalent for H(s)= 1/(S+6). **[D]**
32. Write frequency transformation for BPF and HPF. **[D]**
33. By impulse invariance method obtain the digital filter transfer function and differential equation of the analog filter H(s)=1/S+1**[ID]**

**PART-B**

**REALIZATION**

* + - 1. obtain the direct form –I direct form-II realization of the following system functions.

 y(n)=0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2) **(NOV/DEC-15) [D][7]**

**ANALOG FILTER DESIGN**

1. Explain the procedure for designing analog filter using chebyshev the approximation. **[D][7]**
2. Analyze the design of discrete time IIR filter from analog filter. **(NOV/DEC 2017) [D] [7]**
3. Determine the system function of the lowest order digital Chebyshev filter with the following specifications, 3db ripple in the pass band 0≤w≤ 0.2π and 25db attenuation in the stop band 0.45 π ≤w≤ π. **(APRIL/MAY 2017) [ID][13]**

**IIR FILTER DESIGN BY BILINEAR TRANSFORMATION**

1. Design a digital chebyshev filter to satisfy the constraints

 0.75 ≤│H(ejw)│≤ 1; 0≤w≤ π/2

 │H(ejw)│≤ 0.2; 3π/4 ≤w≤π

 Using bilinear transformation and assuming T=1 Sec. **(MAY/JUNE 2014) [D] [13]**

1. Analyze briefly the different structures of IIR filter**(MAY/JUNE 2014) [ID] [13]**
2. The specification of the desired low pass filter is

 0.8 9≤│H(ejw)│≤ 1.0; 0≤w≤ 0.2π

 │H(ejw)│≤ 0.18; 0.3π ≤w≤π

 Design butterworth filter using bilinear method.**(NOV/DEC 2012) [D] [13]**

4. The specification of the desired low pass filter is

 0.9≤│H(ejw)│≤ 1.0; 0≤w≤ 0.2π

 │H(ejw)│≤ 0.08; 0.4π ≤w≤π Design Butterworth filter using bilinear transformation method.**(NOV/DEC-12) [D] [13]**

5. Design a digital Buttterworth filter satisfying the constraints

 0.707 ≤│H(ejw)│≤ 1; 0≤w≤ π/2

 │H(ejw)│≤ 0.2; 3π/4 ≤w≤π

With T=1sec using bilinear transformation method.**(APRIL/MAY 2015) (APRIL/MAY 2017) [D][13]**

 6. Determine the system function H(z) of the chebyshev low pass digital filter with the specifications.

 ἀp =1 dB ripple in the pass band 0≤w≤0.2π

 ἀs =1 dB ripple in the stop band 0.3 π ≤w≤ π using bilinear transformation T=1sec. **(N0V/DEC-15) [ID][13]**

 7. Convert the given analog filter wih transfer function H(s)= 2/(S+1)(S+2) in to digital filter using bilinear invariant mapping with T=1sec**[D][7]**

 8. Explain the bilinear transform method of IIR filter design. What is warping effect? Explain poles and Zeros mapping procedure clearly. **[ID][7]**

 9. Design a digital chebyshev filter to satisfy the constraints

 0.707 ≤│H(ejw)│≤ 1; 0≤w≤0.2π

 │H(ejw)│≤ 0.1; 0.5 π ≤w≤π

 Using bilinear transformation and assuming T=1 Sec. **(N0V/DEC-14) [D][13]**

**IIR FILTER DESIGN BY IMPULSE INVARIANCE**

 1. Determine the system function of the IIR digital filter for the analog transfer function H(s)= 10/s2+7s+10 with T=0.2 second using impulse invariance method.

 **(NOV/DEC 2014) [D][7]**

2. For the analog transfer function H(s)=2/(S+1)(S+2). Determine H(Z) using impulse invariant method. Assume =1sec. **(N0V/DEC 2014)[D][7]**

 3. The specification of the desired low pass filter is

 0.8 ≤│H(ejw)│≤ 1.0; 0≤w≤ 0.2π

 │H(ejw)│≤ 0.2; 0.32π ≤w≤π

 Design butterworth filter using impulse invariance method.**(NOV/DEC 2013) [D][13]**

 4. The specification of the desired low pass filter is

 1/√2≤│H(ejw)│≤ 1.0; 0≤w≤ 0.25π

 │H(ejw)│≤ 0.24; 0.5π ≤w≤π

 Design chebyshev filter using impulse invariance method.**(NOV/DEC 2012) [D][13]**

 5. Design a digital butterworh filter using impulse invariance method satisfying the constraints. Assume T=1 sec.

 0.8≤│H(ejw)│≤ 1; 0≤w≤0.2π

 │H(ejw)│≤0.2; 0.6≤w≤π**[D] [13]**

 6. Convert the following analog transfer function in to digital using impulse invariant mapping with T=1sec H(s)=3/(S+3)(S+5) **[D][7]**

 7. Design a third order Butterworth digital filter using impulse invariant technique. Assume sampling period T = 1 sec. **(NOV/DEC 2016) [D][13]**

 8**.** Enumerate the steps for IIR filter design by impulse invariance with an example. **(NOV/DEC 2017) [D][6]**

**FILTER DESIGN USING FREQUENCY TRANSLATION**

 1. Convert the single pole low pass filter with system function

 $H\left(z\right)=\frac{0.5\left(1+z^{-1}\right)}{1-0.302z\^-2}$

 into band pass filter with upper and lower cut off frequencies $w\_{u} \& w\_{l}$ respectively. The LPF has 3dB BW of $ w\_{p}=\frac{π}{6}\& w\_{u}=\frac{3π}{4}, w\_{l}=\frac{π}{4} .$ **(NOV/DEC 2016) [ID][13]**

**UNIT-3 FIR FILTER DESIGN**

**PART A**

1. List the different types of structures for realizing FIR system? **(MAY/JUNE-14) [D]**
2. State the properties of FIR filter.**(NOV/DEC-13) [D]**
3. Draw the direct form implementation of the FIR system having difference equation.

y(n)=x(n)-2x(n-1)+3x(n-2)-10x(n-6).**(MAY/JUNE-13) [D]**

1. What are called symmetric and ant symmetric FIR filters? **(MAY/JUNE-2012) [D]**
2. Write the steps involved in FIR filter design. Or What are the techniques of designing FIR filters? **[D]**
3. What are advantages of FIR filter? **[D]**
4. What are the disadvantages of FIR FILTER? **[D]**
5. What is the necessary and sufficient condition for the linear phase characteristic of a FIR filter? **[D]**
6. What is the reason that FIR filter is always stable? **[D]**
7. What condition on the FIR sequence h(n) are to be imposed n order that this filter can be called a liner phase filter? **[D]**
8. What are the features of FIR filter? **[D]**
9. State the advantage and disadvantage of FIR filter over IIR filter. **[D]**
10. What are Gibbs oscillations? **[D]**
11. What is window and why it is necessary? **[D]**
12. Why is window function used in FIR filter design? **(APR/MAY 2015) [D]**
13. Draw a causal FIR filter structure for length m=5. **(APR/MAY 2015) [D]**
14. State the properties of FIR filter. **(NOV/DEC 2015) [D]**
15. Give the desirable characteristics of the window. **(NOV/DEC 2015) [D]**
16. Give the equations specifying Hamming and Hanning window.**(MAY/JUNE 2016) [D]**
17. Realize the following causal linear phase FIR system function: H(z)=2/3+z-1+2/3z-2

**(MAY/JUNE 2016) [D]**

1. What do you understand by linear phase response?**(NOV/DEC 2016)**
2. What are the desirable characteristics of the window? **(NOV/DEC 2016)**
3. What is Gibbs phenomenon? **(APR/MAY 2017) [D]**
4. Compare Hamming window with Blackmann window. **(APR/MAY 2017) [D]**
5. Write the steps involved in FIR filter design. **(NOV/DEC-2017) [D]**
6. Draw the block diagram representation of FIR system. **(NOV/DEC-2017) [D]**
7. What are the desirable properties of windowing technique? **[D]**
8. Write the equation of Hanning window. **[D]**
9. Draw the Direct form I structure of the FIR filter. **[D]**
10. Write the steps involved in FIR filter design. **[D]**
11. Obtain direct cascade realization of the system H(Z) = (1+5Z-1+6Z-2)(1+Z-1) **[D]**
12. Draw a causal FIR filter structure for length L = 5**[D]**

**PART-B**

 **LINEAR PHASE FIR FILTER**

1. Design a linear phase FIR lowpass filter with a cutoff frequency of 0.5π rad/sample by taking 11 samples of ideal frequency response.**[D] [13]**
2. Explain linear phase FIR structure? What are the advantages of such structure? **(MAY/JUNE-16) [D][6]**
3. State and explain the properties of FIR filter? State its importance? **(MAY/JUNE-16) [D][6]**
4. What is linear phase filter? What are the condition to be statisfied by the FIR filter to have linear phase? **(NOV/DEC- 2015) [D][7]**

 **HAMMING WINDOW**

1. The desired frequency response of a filter is

Hd(ejw) ={e-j3w , -π/4≤w≤ π/4

 0 , π/4≤│w│≤π

Determine the filter coefficients of hd(n) using Hamming window with N=7.**(NOV/DEC-16) [D] [13]**

1. Design a high pass filter using hamming window with a cut-off frequency of 1.2 radian/sec and N=9**[D] [13]**
2. Design the HPF with cut off frequency 1.2 radians of length N =9 using hamming window. **[D] [13]**

 **HANNING WINDOW**

1. Design FIR filter with the following desired specification

 Hd(ejw) ={0 , -π/4≤w≤ π/4

 e-j2w , π/4≤│w│≤π

using a hanning window with N=5. .**(NOV/DEC 2017) [D] [13]**

1. Explain the different types of window function. **[7] [D]**
2. Design the HPF with the frequency response

Hd(ejw) ={1 , π/4≤w≤ π

 0 , │w│≤π/4

using Hanning window with N=11. **(APRIL/MAY 2017) [D][13]**

 **RECTANGULAR WINDOW**

1. Design a low pass filter using rectangular window by taking samples N=7 with cut-off frequency of 1.2 rad/sec. **[D] [13]**
2. Explain the principle and procedure for designing FIR filter using rectangular window. **(APRIL/MAY 2015) [D][13]**
3. Using a rectangular window technique design a low pass filter with pass band gain of unity, cut off frequency of 1000 Hz, and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7. **[ID][13]**
4. Design a low pass filter with the cut-off frequency 2$π$ rad/sec using rectangular window with N=7. **(13) [D]**

 **FREQUENCY SAMPLING METHOD**

1. Determine the filter coefficient h(n) of length M=15 obtained by sampling and its frequency response as

 H(2πk/15) = 1 ; k=0,1,2,3,4

 = 0.4 ; k=5

 = 0 ; k=6,7**[D] [13]**

1. Discuss the design procedure of FIR filter using frequency sampling method. **(NOV/DEC 2017)[13] [D]**
2. Design a FIR linear phase digital filter for the response

 Hd(ejw) ={1 , │w│≤ π/6

 0, π/6≤w≤π

 Find the value of h(n) for N=11 plot the frequency response.**(APRIL/MAY-15) [D][13]**

1. Determine the coefficients {h(n)} of a linear phase FIR filter of length M = 15 which has a symmetric unit sample response and a frequency response that satisfies the condition

 H(2πk/15)={ 1 for k=0,1,2,3

 0 for k=4,5,6,7**(APRIL/MAY 2017) [D][13]**

1. Design an ideal BPF with a frequency response

 Hd(ejw) ={1 , π/4≤│w│≤ 3π/4

 0 , otherwise

Find the value of h(n) for N=11 plot the frequency response.**(NOV/DEC-16)[13] [D]**

1. Design a linear phase FIR filter with a cut off frequency of π/2 r /sec .Take N =17 using frequency sampling techniques. **(NOV/DEC-16)[D][13]**

**UNIT-4 FINITE WORD LENGTH EFFECTS**

**PART A**

1. Define over flow error.**(APRIL/MAY-10) [D]**
2. What is overflow oscillations? **(NOV/DEC 2011) [D]**
3. What are limit cycles?**(NOV/DEC 2012) [D]**
4. Compare the digital signal processing systems with fixed point and floating point representation.**(MAY/JUNE 12) [D]**
5. Define signal scaling. Or What is scaling **? [D]**
6. What is truncation? **[D]**
7. How are multiplication and addition carried out in floating point arithmetic?**(MAY/JUNE 2013) [D]**
8. What do you mean by limit cycle oscillations? **(MAY/JUNE 2013) [D]**
9. What is Zero limit cycle oscillations?**(MAY/JUNE 2013) [D]**
10. What is dead band of filter?**(NOV/DEC 2014) [D]**
11. Define finite word length effects.**(NOV/DEC 2014) [D]**
12. Explain briefly quantization noise. **(APR/MAY 2015) [D]**
13. List the types of limit cycle oscillation. **(APR/MAY 2015) [D]**
14. What is meant by fixed point arithmetic? Give example. **(NOV/DEC 2015) [D]**
15. Explain about limit cycle oscillations? **(NOV/DEC 2015) [D]**
16. What are the methods usedto prevent overflow?**(MAY/JUNE-16)**
17. What are the different types of fixed point representation? **(NOV/DEC 2016) [D]**
18. Name the three quantization error due to finite word length registers in digital Filters? **(NOV/DEC 2016) [D]**
19. What are the methods used to prevent overflow? **(APR/MAY 2017) [D]**
20. What is meant by "dead band" of the filter? **(APR/MAY 2017) [D]**
21. Compare the fixed point and floating point number representations. **(NOV/DEC 2017) [D]**
22. What is meant by finite word length effects in digital system ? **(NOV/DEC 2017) [D]**
23. What is product quantization error? **[D]**
24. What are the advantage s of floating point arithmetic? **[D]**
25. Define NTF? **[D]**
26. What is coefficient quantization error? **[D]**
27. What is rounding and what is the range of rounding? **[D]**
28. What is quantization step size? **[D]**
29. Define Noise transfer function? **[D]**
30. What are limit cycles? **[D]**

**PART-B**

**FIXED POINT AND FLOATIONG POINT REPRESENTATIONS**

1. Distinguish between fixed point and floating point arithmetic?**[D][6]**
2. Explain the various formats of the fixed point representation of binary numbers. **(APRIL/MAY 2015) [D][13]**
3. What is meant by finite word length effects on digital filters? List them. **(APRIL/MAY 2015) [D][13]**
4. Consider a second order IIR filter with Find the effect on quantization on pole locations of the given system function in direct form and in cascade form. Assume *b = 3* bits

H(z)=1/(1-0.5z-1)(1-0.45z-1) **[ID] [13]**

**LIMIT CYCLE OSCILLATIONS**

1. Determine the dead band of the system *y(n) = 0.2y(n – 1) + 0.5y(n – 2) + x(n*)

Assume 8 bits are used for signal representation. **[D] [13]**

1. Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation : y(n) = 0.95 y(n-1) + x (n) ; x(n)= 0 and y(n-1)= 13. Determine the dead range of the system. **[D] [13]**
2. Explain the limit cycle oscillations due to product round off and overflow error. **[D][7]**
3. Explain the characteristics of limit cycle oscillations with respect to the system described by the difference equation y(n) = 0.95y(n -1)+ x(n) . **[D] [13]**

Determine the dead band. **(APRIL/MAY 2017) [D][13]**

1. Show the dead band effect on y(n)=0.95y(n-1)+x(n) system restricted to 4 bits. Assume x(0)=0.75 and y(-1)=0**[D] [13]**
2. Study the limit cycle behaviour of the system describe by

 $y\left(n\right)=Q\left[αy\left(n-1\right)\right]+x\left(n\right),$ where y(n) is the output of the filter and

 Q[.] is quantization. Assume $α=\frac{7}{8}, x\left(0\right)=\frac{3}{4} \& x=0, for n>0$

 Choose 4 bit sign magnitude. **(NOV/DEC 2016) [D][13]**

1. Explain the characteristics of limit cycle oscillation with respect to the system

described by the difference equation :

y (n) = 0.95 y (n -1) + x(n); x(n)= 0 and y (-1) = 13. **(NOV/DEC 2017) [D][13]**

1. Define zero input limit cycle oscillation and explain. **(NOV/DEC 2017) [D][13]**

**QUANTISATION NOISE POWER**

1. Find the output round off noise power for the system having transfer function

 H(z)=1/(1-0.5z-1)(1+0.4z-1)which is realized in cascade form.assume word length is 4 bits.(8) **[D] [13]**

1. Derive the equation for quantization noise power. **(MAY/JUNE 2016)[7][D]**
2. Two first order filters are connected in cascaded whose system functions

of the individual sections are H1(z) = 1/(1 -0.5z-1) and H2(z) = 1/(1 -0.6z-1). Determine the overall output noise power. **(APRIL/MAY 2017) [D][13]**

1. Find the output round off noise power for the system having transfer function which is realized in cascade form. Assume word length is 4 bits. **(APRIL/MAY 2015) [D][13]**

**TRUNCATION AND ROUNDING**

1. The coefficients of a system defined by

 H(z) = 1

 (1-0.3z-1) (1-0.65z-1) are represented in a number system with a sign bit and 3 data bits using signed magnitude representation and truncation. Determine the new pole locations for direct realization and for cascade realization of first order systems. **[D] [13]**

1. Discuss in detail the errors resulting from rounding and truncation? **(MAY/JUNE 2016)[7]**

**PRINCIPLE OF SCALING**

1. For the digital network shown in figure find H(z) and scale factor. So to

 avoid over flow register $A\_{1}$ **(NOV/DEC 2016) [D][13]**



**QUANTISATION ERROR**

1. Discuss the following:
2. Product quantization error
3. Limit cycle oscillations**(MAY/JUNE 2016)[13][D]**
4. Explain the quantization process and errors introduced due to quantization. **(NOV/DEC 2017) [D][13]**
5. Discuss various common methods for quantisation. **(NOV/DEC 2015) [D][13]**

**UNIT 5 MULTIRATE SIGNAL PROCESSING**

**PART A**

1. State some applications of DSP? **[D]**
2. Define sampling rate conversion. **[D]**
3. What is decimation? **[D]**
4. Give short note on sub-band coding? **[D]**
5. Give brief note on speech processing? **[D]**
6. Give short note on image enhancement? **[D]**
7. What is interpolation? **[D]**
8. Give short note on adaptive filter? **[D]**
9. Define sampling rate conversion. **[D]**
10. Find the expression for the following multirate systems. **(MAY/JUNE 2012) [D]**
11. Write the need of decimation? **(APR/MAY 2014) [D]**
12. What is decimation and interpolation? **(NOV/DEC 2014) [D]**
13. State the basic operations of multi-rate signal processing. **(APR/MAY 2015) [D]**
14. What are the uses of adaptive filtering? **[D]**
15. Define Oversampling A/D converter. **[D]**
16. Define Oversampling D/A converter. **[D]**
17. What is pipelining? **(APR/MAY 2015) [D]**
18. What is anti-imaging filter? **(NOV/DEC 2015) [D]**
19. Give the application of multi-rate DSP? **(NOV/DEC 2015) [D]**
20. What is decimation and mention its properties? **[D]**
21. Give the steps in multistage sampling rate converter design. **(MAY/JUNE2016) (NOV/DEC- 2013) [D]**
22. Write any 4 applications of multi-rate signal processing. **(MAY/JUNE 2016) [D]**
23. What is the need for anti imaging filter after up sampling signal? **(NOV/DEC 2016) [D]**
24. What is meant by adaptive filter? **(NOV/DEC 2016) [D]**
25. Define adaptive filtering. **(APR/MAY 2017) [D]**
26. Define speech processing. **[D]**
27. List the applications of multi-rate signal processing. **(APR/MAY 2017) (NOV/DEC- 2013) [D]**
28. Write the input output relationship for a decimator. **(NOV/DEC- 2017) [D]**
29. State the applications of adaptive filtering. **(NOV/DEC- 2017) [D]**
30. What are the different techniques for voice compression? **[D]**

**PART B**

**MULTIRATE SIGNAL PROCESSING**

1. Explain in detail the two basic operations in multi-rate signal processing. **(APR/MAY 2015) [D][7]**

**DECIMATION AND INTERPOLATION**

1. Explain the efficient implementation of polyphase decimator and interpolator. **(MAY/JUNE 2016)[13] [D]**
2. Explain the concept of deciation by a factor D and interpolation by factor I. With help of equation explain sampling rate conversion by a rational factor I/D. **(APR/MAY 2017) [D] [13]**
3. Discribe the following:
4. Oversampling A/D converter
5. Oversampling D/A converter. **(NOV/DEC- 2015) [D][7]**
6. Explain the polyphase implementation of FIR filter for decimator and interpolator? **(NOV/DEC- 2015) [D] [13]**
7. The signal x(n) is given by x(n)={0,1,2,3,4,5,6,0,1,2,3……….
8. Obtain the decimated signal by facor 2.
9. Obtain the interpolated signal by factor 2. **(MAY/JUNE 2013)[13] [ID] [13]**

**SAMPLING RATE CONVERSION**

1. i.Explain in about detail the multistage implementation of sampling rate conversion.
	* 1. For the multirate system shown in figure develop an expression for

 the output y(n) as a function of i/p x(n) **(NOV/DEC 2016) [D][13]**

1. i. Show that the upsampler and down sampler are time variant

 systems.

ii.The frequency response of x(n) is shown in figure



 If the input is passed through a down sampler by 2, find the

 frequency response of output and give your comment on aliasing. **(NOV/DEC 2016) [13][ID]**

1. How does the sampling rate increase by an integer factor I ? Derive the

 input-output relationship in both time and frequency domains. **(NOV/DEC- 2017) (MAY/JUNE 2013) [ID][13]**

1. Discuss various applications of multistage sampling rate converter. **(NOV/DEC- 2015) [D][7]**
2. Discuss the steps involved in multistage sampling rate converter. **(MAY/JUNE 2016)[7] [D] [13]**
3. Explain sampling rate conversion by a rational factor and derive input output relation in both time and fr equency domain. . **(MAY/JUNE 2012)[7] [D]**
4. Explain the multistage implementation of sampling rate conversion. . **(MAY/JUNE 2012)[7] [D]**
5. Explain the design of narrow band filter using sampling rate conversion. . **(MAY/JUNE 2012)[7] [D]**
6. Explain the application of sampling rate conversion in sub – band coding. . **(MAY/JUNE 2012)[7] [D]**

**ADAPTIVE FILTER**

1. . Explain the operation of adaptive filter with suitable diagrams and equations. **(APR/MAY 2017) [D] [13]**

**ADAPTIVE FILTER TO EQUALISATION**

1. Discuss in detail about any two applications of adaptive filtering with necessary

Diagrams. **(NOV/DEC- 2017) [D] [13]**

1. Explain the steps in speech coding using transform coding. **(MAY/JUNE 2016)[7] [D]**
2. Draw and explain the block diagram of subband coding system. **(APR/MAY 2015) [D] [13]**
3. Discuss about the musical sound processing. **(APR/MAY 2015) [D] [13]**