CONTENTS

Unit – I

Discrete Fourier Transform

Part – A

1. What is the relation between DTFT and DFT?

Let x(n) be a discrete time sequence.
Now DTFT[x(n)] = X(n) or FT[x(n)] = X(n)
& DFT[x(n)] = X(k).

The X(n) is a periodic continuous function of n and X(k) is an N – point periodic sequence.

The N –point sequence x(k) is actually N samples of X(n) which can be obtained by sampling one period of X(n) at N equal intervals.

2..Distinguish between Discrete Time Fourier Transform and Discrete Fourier Transform.

(Or) Distinguish between DFT and DTFT. [May/June 2010]

DFT	DTFT
1. Obtained by performing sampling operation in both the time and frequency domains.	1. Sampling is performed only in time domain.
2. Used to convert Continuous function of n. to discrete function of n.	2. It is a Continuous function of n.

3. What is the draw back in Fourier Transform and how it is overcome?

The drawback in Fourier Transform is that it is a continuous function of n and so it cannot be processed by digital system. This drawback is overcome by using Discrete Fourier transform. The DFT converts the continuous function of n to a discrete function of n.

4. Write two applications of DFT.

(a)The DFT is used for **spectral analysis** of signals using a digital computer.
(b)The DFT is used to perform l**inear filtering** operations on signals using digital computer.
(c)**Correlation**

5.When an N- point periodic sequence is said to be even or odd sequence.?

[May/June 2011]

An N – point periodic sequence is called even if it satisfies the condition.

$X(n-N) = x(n)$; for $0\leq n\leq (N-1)$

An N – point periodic sequence is called odd if it satisfies the condition.

$X(n-N) = -x(n)$; for $0\leq n\leq (N-1)$

6.. List any four properties of DFT. [May/June 2009]

Let $DFT\{x(n)\} = X(k)$, $DFT\{x_1(n)\} = X_1(k)$ and $DFT\{x_2(n)\} = X_2(k)$

a. **Periodicity:** $X(k+N) =X(k)$; for all k

b**. Linearity:** $DFT\{a_1x_1(n)+a_2\, x_2(n)\} =a_1X_1(k)+a_2X_2(k)$; where a_1 and a_2 are constants.

c. **DFT of time revised sequence**: $DFT\{x(N-m)\} =X(N-k)$

d. **Circular Convolution:** $DFT\ \{x_1(n)©x_2(n)\} = X_1(k)\ X_2(k)$

7. Why linear convolution is important in DSP? [May/June 2008]

The response or output of LTI discrete time system for any input x(n) is given by linear convolution of the input x(n) and the impulse response h(n) of the system. This means that if the impulse response of a system is known, then the response for any input can be determined by convolution operation.

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8. Write the properties of Linear Convolution. [May/June 2008]

a. Commutative Property: $x(n) * h(n) = h(n) * x(n)$

b. Associative Property: $[x(n) * h_1(n)] * h_2(n) = x(n) * [h_1(n) * h_2(n)]$

c. Distributive Property: $x(n) * [h_1(n) + h_2(n)] = [x(n) * h_1(n)] + [x(n) * h_2(n)]$

9.What is Zero Padding? Why it is needed.

Appending Zeros to a sequence in order to increase the size or length of the sequence is called Zero Padding.

In circular convolution, when the two input sequences are different size, then they are converted to equal size by zero padding.

10. What is FFT? [Nov/Dec 2012]

The Fast Fourier Transform is needed to compute DFT with reduced number of calculations. The DFT is spectrum analysis & filtering operations on the signals using digital computers.

11. How many multiplications and additions are required to compute N point DFT using radix-2 FFT?

The number of multiplications and additions required to compute N point DFT using radix-2 FFT are $N \log_2 N$ and $N/2 \log_2 N$ respectively,

12. What is DIT radix – 2 FFT? [May/June 2008]

The Decimation in Time (DIT) radix – 2 FFT is an efficient algorithm for computing DFT. In DIT radix – 2 FFT, the time domain N – point sequence is decimated into 2 – point sequences. This process is continued until we get N – point DFT.

13. What is phase factor or twiddle factor?
The complex number $\mathbf{W_N}$ is called phase factor or twiddle factor. $\mathbf{W_N = e^{-j2\pi/N}}$. It also represents an $\mathbf{N^{th}}$ root of unity.

14. What is DIF radix – 2 FFT? [Nov/Dec 2011]
The Decimation in Frequency (DIF) radix – 2 FFT is an efficient algorithm for computing DFT. In this algorithm the N – point time domain sequence is converted in to two numbers of N/2 point sequences. Here the equations forming N/2 point sequences, N/4 point sequences etc.; are obtained by decimation of frequency domain sequences. This process is continued until we get N – point DFT.

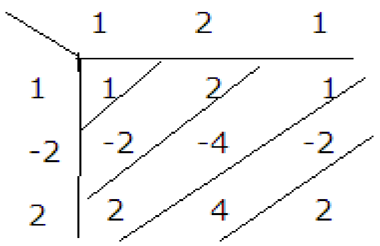
15. Compare the DIT and DIF radix – 2 FFT. [May/June 2006]

What are the differences & similarities between DIT & DIF?

S. No	DIT radix – 2 FFT.	DIF radix – 2 FFT.
1.	The time domain sequence is decimated.	The frequency domain sequence is decimated.
2.	When the input is in bit reversed order, the output will be in normal order and vice versa.	When the input is in bit normal order, the output will be in bit reversed order and vice versa.
3.	In each stage of computations, the phase factors are multiplied before add and subtract operations.	In each stage of computations, the phase factors are multiplied after add and subtract operations.
4.	The value of N should be expressed such that N = 2m and this algorithm consists of m stages of computations.	The value of N should be expressed such that N = 2m and this algorithm consists of m stages of computations.
5.	Total number of arithmetic operations is Nlog2N complex additions and (N/2) log2N complex multiplications.	Total number of arithmetic operations is Nlog2N complex additions and (N/2) log2N complex multiplications.

16.Obtain the circular convolution of the following sequence x(n)={1,2,1}; h(n)={1,-2,2}.

[May/June 2007]



y(n)={1,0,-1,2,2}

17.State the advantages of FFT over DFTs

Efficient computation of DFT
Reduced time required for calculation
Less complex additions
Less complex multiplications
Increased efficiency

18.State any two properties of Discrete Fourier Transform. [Nov/Dec 2011]

1. Linaerity:

If DFT{x(n)} = X(k),
then **DFT $\{a_1x_1(n)+a_2\, x_2(n)\} = a_1X_1(k)+a_2X_2(k)$;** where a_1 and a_2 are constants.

2. Time shifting property:

If DFT{x(n)} = X(k),
then **DFT{x(n-m)} = $X(k)e^{\frac{-j2\pi km}{N}}$**

19.How many stages of decimations are required in the case of a 64point radix 2 DIT FFT Algorithm?[May/June 2012]

The Number of additions required in the computation of 64-point using FFT is $N \log_2 N$ (i.e.) $64 \log_2 64 = 64\,[\log 64 / \log 2] = 384$.

The Number of multiplications required in the computation of 64-point using FFT is $[N/2] \log_2 N$ (i.e.) $[64/2] \log_2 64 = 32\,[\log 64 / \log 2] = 192$.

20.What is meant by in-place computation? [May/June 2013]

An algorithm (DIT/DIF) that uses the same location to store both the input and output sequence is called in-place algorithm. **[OR]**

An "in-place computation "in FFT is simply an FFT that is calculated entirely inside its original sample memory. In other words, calculating an "in place" FFT does not require additional buffer memory.

21.What is meant by bit reversal? [May/June 2011]
The FFT time domain decomposition is usually carried out by a **bit reversal sorting** algorithm. This involves rearranging the order of the *N* time domain samples by counting in binary with the bits flipped left-for-right.

INPUT	INDEX	BINARY REPRESENTATION	BIT REVERSED BINARY	BIT REVERSED INDEX	BIT REVERSED INDEX
x(0)	0	00	00	0	x(0)
x(1)	1	01	10	2	x(2)
x(2)	2	10	01	1	x(1)
x(3)	3	11	11	3	x(3)

22.What is the relationship between DTFT & DFT? [May/June 2008]
a.DFT of a discrete time signal can be obtained by sampling the DTFT of the signal.
b.The drawback in DTFT is that the frequency domain representations of a discrete time signal obtained using DTFT will be a Continuous function of n.
c.The DFT has been developed to convert a continuous function of n to a discrete function of n.
d.The sampling of the DTFT is conventionally performed at N equally spaced frequency points, $0 \le w \le 2\prod$.

23. Compare the number of multiplications required to compute the DFT of a 64 point sequence using direct computation and that using FFT.
The Number of multiplications required in the computation of 64-pointusing FFT is $[N/2] \log_2 N$ (i.e.) $[64/2] \log_2 64 = 32 [\log 64 / \log 2] = 192$.
The Number of multiplications required in the computation of 64-point using DFT is N^2 (i.e.) $64^2 = 4096$.

24.What are the applications of FFT algorithms? [May/June 2006]
The applications of FFT algorithm include
i) linear filtering
ii) correlation
iii) Spectrum analysis

PART-B

1. Check whether the following systems are linear (Nov/Dec 2015)

(1)Y(n)= $\frac{1}{N} \sum_{m=0}^{N-1} x(n-m)$ **(4)**

(2) $Y(n)=[x(n)]^2$ (4)

2. **Find the impulse response of the sytem y(n)-y(n-1)=x(n)+x(n-1) (Nov/Dec 2015)**
3. **Find the DFT of x(n)={1,1,0,0}.(MAY 2012)**
4. **Determine the N-point DFT of the following sequences.**

   a.X (n) = δ (n)

   b.X (n) = δ (n - n_0). (DEC 2011)
5. **(i) Determine the 8-point DFT of the sequence x(n) = { 0, 0, 1, 1, 1, 0, 0, 0}. (Apr/May 2015)**
6. **State the following properties of DFT. (MAY 2013) (8 mark)**

   a.time reversal

   b.parsavel's theorem
7. **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 (16) (Nov/Dec 2016)**
8. **(i) Explain decimation in time FFT (DITFFT) algorithm for 8 point DFT computation. (MAY 2012)**
9. **Compute the DFT for the sequence {1,2,3,4,4,3,2,1} using radix -2 DIF FFT algorithm(16) (May/June 2017)**
10. **What is decimation in frequency algorithm? Write the similarities and differences between DIF and DIT algorithms. Apr/May 2015 (or) Illustrate the reduction of an 8-point DFT to two 4-point DFT's by decimation in frequency.(Nov/Dec 2015)**
11. **Compute IDFT of the sequence X(K)={7,-0.707-j0.707,-j,0.707-j0.707,1,0.707+j0.707,-0.7.07+j0.707} using DIF algorithm(10) (Nov/Dec 2016)**
12. **(i) Compare the computational complexity of direct DFT computation and FFT computation of a sequence with N=64(DEC ' 09)(4 mark)**

    (ii) Prove that FFT algorithms help in reducing the number of computations involved in DFT computation. (DEC 2012) (8 mark)

13. **(i)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.(16) (April/May 2017)**

    (ii) Compute the 8 point circular convolution

    a. **$x_1(n)$={1,1,1,1,0,0,0,0}**

b. $x_2(n)=\sin\frac{3\pi n}{8}$**, $0 \le n \le 7$ using matrix method(12) (May/June 2016)**

14. With appropriate diagrams discuss how overlap-save method and overlap – add method are used.(16) Apr/May 2015 (or)State the difference between (a) overlap-save (b) overlap-add (4) (May/June 2016)

15. (i) Perform the linear convolution of finite duration sequences h(n)={1,2} and x(n)={1,2,-1,2,3,-2,-3,-1,1,2,-1} by overlap save method (6) (Nov/Dec 2016)
(ii)compute 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 add method. (DEC 2011) (DEC 2014) (16 mark)

16. perform the linear convolution of the given sequences x(n) = {1,-1,1,-1),h(n) = {1,2,3,4} using DFT method. (MAY 2013) (8 mark)

Unit – II

IIR Filter Design

Part – A

1. What is frequency warping in Bilinear transformation? [Nov/Dec 2011]

The mapping of frequency from Ω to ω is approximately linear for small value of Ω & ω. For the higher frequencies, however the relation between Ω & ω becomes highly non-linear. This introduces the distortion in the frequency scale of digital filter relative to analog filter. This effect is known as warping effect.

2. What are methods used to convert analog to digital filter?

Approximation of derivatives, Impulse invariant method & Bilinear transformation method.

3. Write the pole mapping rule in Impulse invariant method?

A pole located at $s = s_p$ in the s plane is transferred into a pole in the z plane located at $Z = e^{s_pT_s}$. Each strip of width 2π/T on left half of s-plane should be mapped to region inside the unit circle in z-plane.

The imaginary axis of s-plane is mapped to unit circle in z-plane. Left half of s-plane is mapped to outer region of unit circle.

4. What are the disadvantages of Impulse invariant method?

It provides many to one pole mapping from s-plane to z-plane. aliasing will occur in IIT.

5. What are the advantages of Bilinear transformation method?

The Bilinear transform method provides non linear one to one mapping of the frequency points on the jw axis in the S plane to those on the unit circle in the Z plane.i.e Entire jw axis for $-\infty < w < \infty$ maps uniquely on to a unit circle $-\pi/T < \omega/T < -\pi/T$. This procedure allows us to implement digital high pass filters from their analog counter parts. No aliasing effects.

6. Define prewarping or prescaling. [May/June 2012]

For large frequency values the non linear compression that occurs in the mapping of Ω to w is more apparent .This compression causes the transfer function at high Ω frequency to be highly distorted when it is translate to the w domain. This compression is being compensated by introducing a prescaling or prewarpping to Ω frequency scale. For bilinear transform Ω scale is converted into Ω^* scale (i.e) $\Omega^* = 2/Ts \tan(\Omega Ts/2)$(prewarped frequency)

7. Comparison of analog and digital filters. [May/June 2015]

A Analog filter	DDigital filter
nalog filter both input and output tinuous time signal	digital filter,both the input and output are te time signals.
can be constructed using active assive components.	can be constructed using adder, multiplier elay units.
ese filters operate in infinite frq. e, theoretically but in practice it is d by finite max. operating freq. nding upon the devices used.	q. range is restricted to half the sampling and it is also restricted by max. utational speed available for particular ation.
defined by linear differential eqn.	defined by linear difference eqn

8. What are the advantages of digital filter?
1. Filter coefficient can be changed any time thus it implements the adaptive future.
2. It does not require impedance matching between input and output.
3. Multiple filtering is possible. 4. Improved accuracy, stability and dynamic range.

9. What are disadvantages of Digital Filter?
The bandwidth of the filter is limited by sampling frequency.The performance of the digital filter depends on the hardware used to implement the filter.
The quantization error arises due to finite word length effect in representation of signal and filter coefficient.

10. What is the difference between Chebyshev Filter type I and typeII?
Filter TypeI:
It is all pole filter and exhibits equiripples in the pass band and monotonic characteristics in the stop band.
Filter Type II:It contains both poles and zeros and exhibits a monotonic behaviour in the pass band and equiripple in the stop band.

11. What are the properties of chebyshev filter?
1. For ω ≥ 1 H(jω) decreases monotonically towards zero.
2. For ω ≤ 1 H(jω) it oscillates between 1 and 1\(1+ε^2)

12**. Compare Butterworth filter and chebyshev filter. [May/June 2011]**
<u>Butterworth filter</u>
1.The Magnitude response of Butterworth filter decreases monotonically as the frequency increases.
2. The Transition width is more
3.The order of butterworth filter is more, thus it requires more elements to construct and is expensive.
4. The Poles of the butterworth filter lies along the circle.
5. Magnitude response is flat at ω=0 thus it is known as maximally flat filter.
<u>Chebyshev Filter</u>
1. The Magnitude response of Chebyshev filter will not decrease monotonically with frequency because it exhibits ripples in pass band or stop band.
2. The Transition width is very small
3. For the same specifications the order of the filter is small and is less complex and

inexpensive.
4. The poles of chebyshev filter lies along the ellipse.
5. Magnitude response produces ripples in the pass band or stop band thus it is known as equripple filter.

13. What are the properties of Chebyshev filter?

1. The magnitude response of the Chebyshev filter exhibits ripples either in the pass band or in the stop band. 2.The poles of a Chebyshev filter lie on an ellipse.

14. What are the different structures for realization of IIR systems?

Direct Form I, Direct Form II, Cascade, Transposed, Parallel, Lattice ladder Structures

15. What is Butterworth approximation?

The frequency response characteristic of the low pass butterworth filter is monotonic in both pass band and stop band. The response approximate to the ideal response as the order N of the filter increases (flat characteristics).

16. What is the relation between Analog and digital frequency in IIT?

The relation between Analog and digital frequency is given by digital frequency = ΩT
Where Ω = analog frequency and T= sampling period.

17. State the two advantage of bilinear transformation.

It avoids aliasing in frequency components.
The transformation of stable analog filter results in a stable digital filter.

18. What are the parameters (specifications)of a Chebyshev filter?

Pass band ripple, pass band cut off frequency, stop band cut off frequency, attenuation beyond stop band frequency.

19. What is Chebyshev approximation?

In Chebyshev approximation, the error is defined as the difference between the ideal brickwall characteristic and the actual response and this is minimized over a prescribed band of frequencies.

20. Mention the important features of IIR filters.

The physically realizable IIR filter does not have linear phase.
The IIR specifications include the desired characteristics for the magnitude response only

21. What is bilinear transformation? What is the main advantages and disadvantages of this technique?

It is conformal mapping which utilize prewarping technique used to design IIR filters. It is one to one mapping. The relation between analog and digital frequency is nonlinear, ie $\Omega=2/T \tan(\omega/2)$
Advt.: No aliasing effects Dis-advt.: Due to nonlinear relation between ω and Ω distortion occurs in frequency domain of digital filter.

PART – B

1. **Explain in detail butterworth filter approximation. (May 11) (16 mark)**
2. **An analog filter has the following system function. Convert this filter into a digital filter using backward difference (approximative derivative) for the derivance**
   $H(s) = 1 / (s+0.1)^2+9$
3. **If $H_a(s)= 1/ (s+1)(s+2)$ find the corresponding H(Z) using impulse invariant method for sampling frequency of 5 samples /second.(May 16)**
4. **(i)Convert the analog filter into a digital filter whose system function is $H(s) = s+0.2 / (s+0.2)^2+9$. Use impulse invariance technique. Assume T= 1 sec.(Dec 2015)**
   (ii)convert the analog filter with system function $Ha(s) = s+0.1 / (s+0.1)^2+16$ into a digital IIR filter by means of the bilinear transformation. The digital filter is to have a resonant frequency of $\omega r = \pi/4$. (10) Apr/May 2015
5. **Design a third order butterworth digital filter using impulse invariant technique. Assume sampling period T = 1 sec.(Dec 2016)**
6. **design a digital Butterworth filter using impulse invariance method satisfying the constraints. Assume T = 1sec. (DEC 2011) (16 mark)**
   $0.8 \le |H(e^{j\omega})| \le 1; \quad 0 \le \omega \le 0.2\pi$
   $|H(e^{j\omega})| \le 0.2; \quad 0.6\pi \le \omega \le \pi$
7. **design a digital butterworth filter satisfying the following specifications (Dec'07) (16 mark)**
   $0.7 \le |H(e^{j\omega})| \le 1$ for $0 \le \omega \le 0.2\pi$
   $|H(e^{j\omega})| \le 0.004$ for $0.6\pi \le \omega \le \pi$
   assume$T = 1$Ssec. Apply impulse-invariant transformation

8. **Explain the bilinear transform method of IR filter design. What is warping effect? Explain the poles and zeros mapping procedure clearly. (May 11) (16 mark)**
9. **show that a stable analog filter is mapped to a stable digital filter using bilinear transform (May'09)**
10. **Explain in details the steps involved in the design of IIR filter using bilinear transformation.(16) Apr/May 2015**
11. **Determine the system function H(z) of the chebyshev´s low pass digital filter with the specification $\alpha p=1$ db ripple in the pass band $0 \le \omega \le 0.2\pi$, $\alpha s=15$ db ripple in the stop band $0.3\pi \le \omega \le \pi$ Using bilinear transformation (assume T= 1sec). (Dec 10)**

12. **Design a Butterworth filter using the bilinear transformation for the given specification. (MAY 2012)**
    $0.8 \le H(e^{i\omega}) \le 1 \quad 0 \le \omega \le 0.2\pi$
    $H(e^{i\omega}) \le 0.2 \quad 0.6\pi \le \omega \le \pi$

13. Design a digital second order low – pass Butterworth filter with cut-off frequency 2200 Hz using bilinear transformation. Sampling rate is 8000 Hz. (DEC 2012)

14. Write down the steps to design digital filter using bilinear transform technique and design using a HPF with a pass band cutoff frequency of 1000Hz and down 10db at 350 Hz the sampling frequency is 5000Hz.(May 2016)

15. Dertermine the system function of the lowest order digital chebyshev filter with the following specifications, 3db ripple in the pass band and $0 \le |\omega| \le 0.2\pi$ and 25db attenuation in the stopband $0.45\pi \le |\omega| \le \pi$ (May 17)

16. (i)Obtain the direct from I, direct form II and cascade form realization of the following system functions. (DEC 2011) (16 mark)

$$Y(n) = 0.1\, y(n-1) + 0.2\, y(n-2) + 3x(n) + 3.6\, x(n-1) + 0.6\, x(n-2).$$

(ii)Obtain the direct form I,direct form II, cascade and parallel form realization for the system

$y(n) = -0.1\, y(n-1) + 0.2\, y(n-2) + 3x(n) + 3.6\, x(n-1) + 0.6\, x(n-2)$. (Dec 10) (16 mark)

17. Obtain direct form I and direct form II realization of $H(z) = \dfrac{1 + 2z^{-1}}{1 - 1.5z^{-1} + 0.4z^{-2}}$ (MAY 2012)

18. Determine the cascade form and parallel form implementation of the system governed by the transfer function (DEC 2012)

$$H(Z) = \frac{(1+Z^{-1})(1-5Z^{-1}-Z^{-2})}{(1+2Z^{-1}+Z^{-2})(1+Z^{-1}+Z^{-2})}$$

19. Draw the structure for the IIR filter in direct form – II for the following transfer function.(6) Apr/May 2015

$$H(Z) = \frac{(2+2Z^{-1})(4+2Z^{-1}+3Z^{-2})}{(1+0.6Z^{-1})(1+Z^{-1}+0.5Z^{-2})}$$

20. Explain the procedure for designing analog filters using the chebysher approximation. (DEC 2012) (8 mark)

Unit – III

FIR FILTER DESIGN

PART – A

1. List out the advantages and disadvantages of FIR filters.

Advantages:

- **L**inear phase FIR filter can be easily designed.
- Efficient realization of FIR filter exists as both recursive and non-recursive structures.
- FIR filter realized non-recursively stable.

Disadvantages:

The duration of impulse response should be large to realize sharp cutoff filters. The non integral delay can lead to problems in some signal processing applications.

2. **State the properties of FIR filters.**

1. FIR filter is always stable.
2. A realizable filter can always be obtained.
3. FIR filter has a linear phase response.

3. **Compare FIR and IIR filters?**

FIR filters	**IIR filters**
1. Impulse response is of finite duration.	Impulse response is of infinite duration.
2.These filters can be easily designed to have perfectly linear phase	These filters do not have linear phase.
3.It is implemented using structure with no feedback.	It is implemented using structure having feedback.
4.It is a non-recursive system.	It is a recursive system.
5.It has all zero"s, E.g: $h(n) = \begin{cases} 2\,; & \|n\| \le 4 \\ 0\,; & otherwise \end{cases}$	Both poles and zero"s E.g: $h(n) = a^n u(n); n \ge 0$

4. What condition on the FIR sequence h(n) is to be imposed in order that this filter cane be called a linear phase filter?

The conditions are (i) Symmetric condition h(n) = h(N-1-n)
(ii) Antisymmetric condition h(n) = -h(N-1-n)

5. State the conditions for a digital filter to be causal and stable.
A digital filter is causal if its impulse response h(n) = 0 for n < 0
A digital filter is stable if its impulse response is absolutely summable

$$\sum_{n=-\infty}^{\infty} |h(n)| < \infty$$

6. What do you understand by linear phase response.
The necessary and sufficient condition for linear phase characteristics in FIR filter is the impulse response h(n) of the system should have the symmetry property,i.e, H (n) = h(N-1-n)
where N is the duration of the sequence.

7. **What are the desirable characteristics of windows?** The desirable characteristics of the window are
1. The central lobe of the frequency response of the window should contain most of the energy and should be narrow.
2. The highest side lobe level of the frequency response should be small.
3. The side's lobes of the frequency response should decrease in energy rapidly as w tends to π.

8. **Why FIR filters are always stable?**
FIR filters are always stable because all its poles are at the origin

9. Define Gibbs phenomenon.
One possible way of finding an FIR filter that approximates H(ejw) would be to truncate the infinite Fourier series at n= ± (N-1/2).Abrupt truncation of the series will lead to oscillation in the passband and stop band. This is known as Gibbs phenomenon,

10. What is need for windowing technique for design FIR filter.(or) what is window and why its necessary.
One possible way of finding an FIR filter that approximates H(ejw) would be to truncate the infinite Fourier series at n= ± (N-1/2).Abrupt truncation of the series will lead to oscillation in the passband and stop band. These oscillations can be reduced through the use of less abrupt truncation of the fourier series. This can be achieved by multiplying the infinite impulse response by a finite weighing sequence w(n) called window.

11. What are the disadvantages of Fourier series method?
In designing FIR filter using Fourier series method the infinite duration impulse response is truncated at n= ± (N-1/2)..Direct truncation of the series will lead to fixed percentage overshoots and undershoots before and after an approximated discontinuity in the frequency response.

12. **Write the equation of hamming windows of function.**

W(n)=0.54-0.46cos(2πn/N-1) for0≤n≤N-1

13. **Compare the rectangular window , Hanning window and hamming window**.

S.No	Rectangular window	Hanning Window	Hamming Window
1	The width of main lobe in window spectrum is $4\pi/N$	The width of main lobe in window spectrum is $8\pi/N$	The width of main lobe in window spectrum is $8\pi/N$
2	The maximum side lobe magnitude in window spectrum is –13dB.	The maximum side lobe magnitude in window spectrum is –31dB.	The maximum side lobe magnitude in window spectrum is –41dB
3	In window spectrum the side lobe magnitude slightly decreases with increasing w	In window spectrum the side lobe magnitude decreases with increasing w.	In window spectrum the side lobe magnitude remains constant
4	In FIR filter designed using rectangular window the minimum stop band attenuation is 22dB	In FIR filter designed using hanning window the minimum stop band attenuation is 44dB	In FIR filter designed using hamming window the minimum stop band attenuation is 44dB

14. For What type of filters frequency sampling method is suitable?
Frequency sampling method is attractive for narrow band frequency selective filters where only a few of the samples of the frequency response are non-zero.

15. **What is the drawback in FIR filter design using windows and frequency sampling method? How is it overcome?**
The FIR filter designs by window and frequency sampling method do not have precise control over the critical frequencies such as ω_p, and ω_s.
This drawback can be overcome by designing FIR filter using Chebyshev approximation technique. In this technique an error function is used to approximate the ideal frequency response, in order to satisfy the desired specification.

16. Write the procedure for designing FIR filter using frequency-sampling method.

- Choose the desired (ideal) frequency response Hd(w).
- Take N-samples of Hd(w) to generate the sequence
- Take inverse DFT of to get the impulse response h(n).

The transfer function H(z) of the filter is obtained by taking z-transform of impulse response

17. 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).

PART- B

1. **List the steps involved by the general process os designing a digital filter.(Dec 2015) [Explain design procesdure for IIR and FIR filters]**
2. **Explain the principle and procedure for designing FIR filter using rectangular window.(16) Apr/May 2015**
3. **Discuss the window method of designing FIR filters in detail. (May '10) (6 mark)**
4. **Explain the designing of FIR filters using windows. (May 11) (16 mark)**
5. **Design an ideal high pass filter using hanning window with a frequency response.**
   $H_d(e^{j\omega}) = 1;\ \pi/4 \le |\omega| \le \pi$
   $= 0;\ |\omega| \le \pi/4$. Assume N=11. (DEC 2011) (16 mark)
6. **Design a FIR low pass filter having the following specification using hanning window. (MAY 2012) (16 mark) Apr/May 2015**
   $H_\alpha(e^{i\omega}) = 1$ for $-\pi/6 \le |\omega| \le \pi/6$ And N=7.
   0 for otherwise
7. **Design an ideal high pass filter with a frequency response**
   $Hd(e^{j\omega})=\{1$ for $\pi/4 \le |\omega| \le \pi$
   0 for $|\omega| \le \pi/4$
   Find the values of h(n) for N=11 using hanning window. Find H(z) and determine the magnitude response. (May 2017) (16 mark)
8. **Design a ideal BPF with a frequency response**
   $Hd(e^{j\omega})=\{1$ for $\pi/4 \le |\omega| \le \pi$
   0 for $|\omega| \le \pi/4$
   Find the values of h(n) for N=11, and plot the frequency response.(Dec 2016)
9. **Design a digital FIR band pass filter with lower cut – off frequency 2000 Hz and upper cut off frequency 3200 Hz using hamming window of length N = 7.sampling rate is 10000 Hz. (DEC 2012) (8 mark)**

10. **design a FIR filter with the following desired specification**
    $Hd(e^{j\omega}) = \{0,\ -\pi/4 \le \omega \le \pi/4$
    $e^{-j2\omega},\ \pi/4 \le |\omega| \le \pi$
    using a hanning windows with N=5. (MAY 2013) (16 mark)
11. **the desired response of a filter is**
    $H_d(e^{j\omega}) = e^{-j3\omega}\quad -\pi/4 \le \omega \le \pi/4$
    $= 0\quad \pi/4 \le \omega \le \pi$
    Design the filter for M=7 using hamming window. (may 2016)(16 mark)
12. **Design an ideal differentiator with frequency response.**
    $H(e^{j\omega}) = j\omega;\ -\pi \le \omega \le \pi$ using hamming window with N = 8. (DEC 2011) (16 mark)
13. **(i)Explain the frequency sampling method of FIR filter design. (MAY 2012) (8 mark)**
    (ii)Discuss the design procedure of FIR filter using frequency sampling method. (MAY 2013) (8 mark)

14. 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

$H_r(2\pi k/15)$={1, for k=0,1,2,3

0, for k=4,5,6,7 (Dec 10) (May 2017)

15. design an FIR low pass digital filter by using the frequency sampling method for the following specifications(DEC 2012) (16 mark)
Cutoff frequency = 1500Hz
Sampling frequency = 1500HZ
Order of the filter : N=10
Filter length required L = N+1=11. (That is N = 11)

16. design a linear phase FIR filter with a cut off frequency of π/2 r/sec. Take n = 17 using frequency sampling techniques(Dec 2016)

17. determine the direct form realization of system function (MAY 2012) (8 mark)
$H(z) = 1+2z^{-1} - 3z^{-2} -4z^{-3} + 5z^{-4.}$

18. Determine the frequency response of FIR filter defined by y(n)=0.25x(n)+x(n-1)+0.25x(n-2).Calculate the phase delay and group delay. (MAY 2013) (8 mark)

19. Realize the system function $H(z) = (2/3)Z+1+(2/3)Z^{-1}$ by linear phase FIR structure. (May 11) (16 mark)

Unit – IV

FINITE WORD LENGTH EFFECTS

PART – A

1. What is meant by fixed point arithmetic? Give example.

In fixed point arithmetic the position of binary point is fixed. The bit of the right represents the fractional part of the number and those to the left represent the integer part.

2. Compare the fixed point and floating point arithmetic.

s.no	Fixed point arithmetic	Floating point arithmetic
1.	Fast operating	Slow operating
2.	Relatively economical	Move expensive because of costlier hardware
3.	Round off errors occur only for addition.	Round off errors occurs with both addition and multiplication
4.	Quantization errors occur only in multiplication.	Quantization errors occur in both addition and multiplications.
5.	The accuracy of the result is less due to smaller dynamic range	The higher due accuracy of the result will be higher due to the larger dynamic range
6.	Overflow occurs in addition	Overflow does not arise
7.	Used in small computer	Used in larger & general purpose computer

3. What are advantages of floating point arithmetic?

1. The accuracy of the result will be higher due to the larger dynamic range
2. Overflow does not arise.

4. What are the different types of fixed point representation.

1.Sign magnitude form
2. 1's complement form
3. 2's complement form

5. Express the fraction 7/8 and -7/8 in sing magnitude, 2 is complement and 1 is complement.

Fractions	Sign magnitude	1's compliment	2's compliment
+7/8	0.111	0.111	0.111
-7/8	1.111	1.000	1.001

6. Express the fraction (-7/32) in signed magnitude and two´s complement notations using 6 bits.

Fractions	Sign magnitude	1's compliment	2's compliment	
-7/32	1.001	1.110	1.111	

7. Express the fraction (-9/32) in signed magnitude and two´s complement notations using 6bit.

Fractions	Sign magnitude	1's compliment	2's compliment
-9/32	1.111	1.000	1.001

8. What are the two types of quantization employed in digital system? (or) What are the different quantization methods?

1.Truncation
2. Rounding

9. Compare truncation with rounding errors.

Truncation	**Rounding errors**
It is the process of reducing the size of the binary numbers (or) numbers of bits in a binary numbers by discarding all bits less significant .than least significant bit is ret aired	Rounding of a number of b bits is accomplished by choosing the rounded result as the b bit number closer to the original number unrounded
e.g. truncation the following number from 7 bits to 4 bits ,we get 0.0011001 to 0.0011 0.0100100 to 0.0100	e.g. 0.11010 ⇒ 0.111 (3 bits)

10. What is truncation? (or) what does the truncation of data result in.

It is the process of reducing the size of the binary numbers (or) numbers of bits in a binary numbers by discarding all bits less significant .than least significant bit is ret aired

E.g. truncation the following number from 7 bits to 4 bits, we get

1. 0.0011001 to 0.0011, 2. 0.0100100 to 0.0100

11. List the representation for which truncation error is analyzed.
What are the quantization errors due to finite word length registers in digital filters? (or) What are the three types of quantization error occurred in digital systems?

The following errors arise due to the quantization of numbers

1.Input quantization error
2.Product quantization error
3.Co-efficient quantization error

12. What is product quantization error?

Product quantization errors arise at the output of the multiplier. Multiplication of a b bit data with a b bit coefficient results a product having 2b bits. Since a b bit registers is used, the multiplier output must be rounded or truncated to b bits which produces the error.

13. What is the effect of quantization on pole locations?
Ans :Due to quantization on pole locations will be changed in such a way that the system may drive into instability.

14. What are the various factors which degrade the performance of digital implementation when finite word length is used?
Ans: (i) errors due to quantization of input data.
(ii) errors due to quantization of filter coefficients.
(iii) errors due to rounding the product in multiplications.
(iv) limit cycles due to product quantization and overflow additions

15. How would you relate the steady-state noise power due to quatization to the 'b' bits representing the binary sequence?
Ans : $64\ x^2_{rms} / 2^{2b}$

16. What is known as overflow oscillations?
In addition to limit cycle oscillation causing by rounding the result of the multiplication, there are several types of limit cycle oscillation caused by addition which make the filter output oscillate between maximum and minimum amplitudes. Such limits cycles have been referred to as oscillation

17. **What are limit cycle oscillations?**
When a stable IIR filter is excited by a finite input sequence, that is constant, the output will ideally delay to zero. However, the nonlinearities due to the finite precision arithmetic operation often causes periodic oscillation occur in the output. Such oscillation in recursive system called zero input limit cycle oscillation.

18. What is dead band of a filter?
The limit cycle occur as the result of the quantization effects in multiplication. The amplitudes of the output during a limit cycle are confirmed to the range of values that is called dead band of the filter

Dead band = $\frac{1}{2}\ 2^{-b}$ **/** $1-\alpha$

19. What is the need for signal scaling?
The saturation arithmetic eliminate limit cycles due to overflow, but it causes undesirable signal distortion due to the nonlinearity of the clipper. In order to limit the amount of nonlinear distortion, it is important to scale the input signal and the unit sample response between the input and any internal summary node in the system , such that the overflow becomes a rare event

20. What are the methods used to prevent overflow.
1. Signal scalling

PART-B

1. **What is meant by finite word length effects on digital filters? List them.(8) Apr/May 2015**
2. **Explain the various formula of the fixed point representation of binary numbers.(8) (Apr/May 2015)**
3. **Represent the following numbers in floating point format with five bits for mantissa and three bits for exponent. (MAY 2013) (8 mark)**
   (1) 7_{10} (2) 0.25_{10} (3) -7_{10} (4) -0.25_{10}
4. **What is quantization of analog signals? Derive the expression for the quantization error. (DEC 2012) (8 mark)**
5. **Explain truncation and rounding. (MAY 2012)**
6. **(i)What is called quantization noise? Derive the expression for quantization noise power. (MAY 2012) (8 mark)**
7. **(i)explain product quantization error (MAY 2012)**
   (ii)Draw the product quantization noise model of second order IIR system. (MAY 2013
8. **Explain coefficient quantization in IIR filter. (DEC 2012) (8 mark)**
9. **Explain how reduction of product round-off error is achieved in digital filters. (May 11) (8 mark) (16 mark)**
10. **Explain Over flow limit cycle oscillation(DEC 2011) (8 mark) (page 58)**
11. **(i)How to prevent limit cycle oscillations? Explain. (DEC 2012) (8 mark)(also explain signal scalling)**
    (ii)Explain how signal scaling is used to prevent overflow limit cycle in the digital filter implementation with an example. (MAY 2013) (8 mark)
12. **The output of ADC is applied to a digital filter with system function H(z) = 0.5Z / Z-0.5 . Find the output noise power from digital filter when the input signal is quantized to have8 bits.(Dec 2015))**
13. **(i) Consider the transfer function H(z)=H1(z)H2(z) where**
    $H_1(Z) = 1/ 1- \alpha_1 Z^{-1}$, $H_2(Z) = 1/ 1- \alpha_2 Z^{-1}$ and $\alpha_1 = 0.5$ and $\alpha_2 = 0.6$.(May 16)
    (ii) Two first order filters are connected in cascade whose system functions of the individual sections are $H_1(Z) = 1/ 1- 0.5Z^{-1}$, $H_2(Z) = 1/ 1- 0.6Z^{-1}$. Determine the overall output noise power.(may 17)
    (iii) Find the output round off noise power for the system having transfer function H (z) = 1/ (1-0.5 z-1) (1-0.45z-1) which is realized in cascade form. Assume word length is 4 bits.(16) (Apr/May 2015)
14. **Consider a second order IIR filter with**
    H (z) = 1.0 / (1-0.5 z-1) (1-0.45z-1)
    Find the effect on quantization on pole locations of the given system function in direction form and in cascade form. Assume b = 3 bits. (DEC 2011) (8 mark)

15. **an 8-bit ADC feeds a DSP system characterized by the following transfer function $H(Z)= 1/ z+0.5$ Estimate the steady state quantization noise power at the output of the system (Dec'07) (8mark)**

16. **realize the first order transfer function(z)=$1/1-aZ^{-1}$ and draw its quantization model.find the steady state noise power due to product round off. (May'09)**

17. **(i)For a second order digital filter $H(z)=1 / 1- 2r\cos\theta Z^{-1} + r^2 Z^{-2}$. draw the direct form II Realization and find the scale factor So to avoid overflow (May'09)**
(ii)Draw the quantization noise model for a second order systemH(z) = $1/ 1-2r \cos\Theta z^{-1} + r^2 z^{-2}$ and find the steady state output noise variance. (DEC ' 09)(May 16) (8 mark)

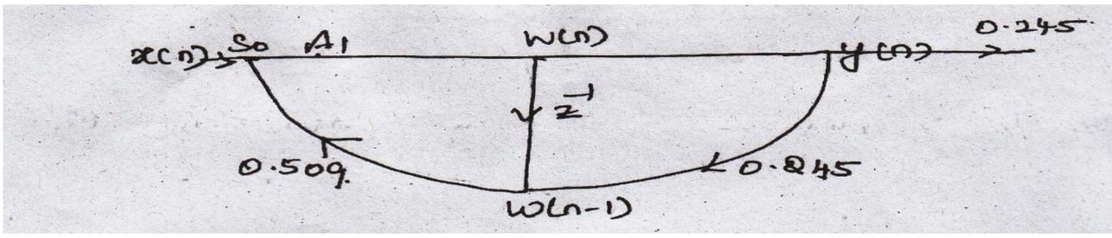
18. **Determine the dead band of the system y(n)=0.2y(n-1)+0.5y(n-2)+x(n).Assume 8 bits are used for signal representation. (MAY 2013) (8 mark)**

19. **A digital system is characterized by the difference equation**
1. **Y(n)=0.9y (n-1)+x(n)**
ii. **With x[n]=0 and initial condition y(-1)=12.**
iii. **Determine the dead band of the system.**

20. **A digital system is characterized by the difference equation Y(n)=0.9y (n-1)+x(n) With x[n]=0 and initial condition y(-1)=12.Determine the dead band of the system**

21. **Study the limit cycle behavior of the system described by y(n) = Q[α y(n-1)] + x(n), where y(n) is the output of the filter and Q[.] is quantization. Assume α= 7/8, x(0) = ¾, x=0, for n>0 choose 4 bit sign magnitude.(DEC 16).**

22. **For the digital network shown in below figure find H(z) and scale factor. So to avoid over flow register A1.**



Unit – V

DSP APPLICATIONS

PART – A

1.What is Echo cancellation?
Echo cancellation is the process of removing echo from a voice communication.

2.What are narrowband filters?
A common need in electronics and DSP is to isolate narrow band of frequencies from a wider bandwidth signal. For example you want to eliminate 60Hz interference in a instrumentation system or isolate the signaling tones in a telephone network. Two types of frequency responses are available. (i) Narrowband Low pass filter (ii) Narrowband Band pass filter.

3.What is decimator or down sampling
The process of reducing the sampling rate by a factor D (down sampling by D) of a signal is called decimation.(i.e sampling rate expansion).

4.What is interpolator or up sampling?
The process of increasing the sampling rate by a factor I (up sampling by I)of a signal is interpolation. (i.e sampling rate expansion).

5.What is sampling rate conversion?
The process of converting the signal from given sample rate to a different sample rate is called sampling rate conversion.

6.What are the drawbacks in multi stage implementation?
(i) While converting digital signal x(n) in analog, D\A converter introduces distortion in signal rebuilding.
(ii) A/D converter gives quantization error

7.What do you mean by sub band coding? (OR) Define sub-band coding.
Sub band coding is a method where speech signal is sub-divided into several frequency bands and each band is digitally encoded separately by allocating different bits per sample to the signal of different sub bands.

8.What is the need for multirate signal processing?
In telecommunication systems, different types of signals have to be processed at a different rate. The system that employs multiple sampling rates in the processing of different signal is known as multirate signal processing system.

9.What is meant by adaptive equalization?
In digital communication system the bandwidth of channel has to be used efficiently. The requirement is to design a reliable system with data to be transmitted at a higher rate. The factors that affect the data in the channel are inter symbol interference and thermal

noise. Adaptive equalizer is used for compensating the channel distortion so that the detected signal is reliable.

10.State the basic operations of Multirate signal processing?
The two basic operations in multirate signal processing are decimation and interpolation. Decimation reduces that sampling rate, whereas interpolation increases the sampling rate.

11.What is the effect of Downsampling on the spectrum of a signal?
Downsampling or decimation is used to avoid aliasing effect.

12.What is an anti-imaging filter?
The filter which is used to remove the image spectra is known as anti-imaging filter.

13.What are the methods are sampling rate conversion of a digital signal? (OR) Give the steps in multistage sampling rate converter design.
Sampling rate conversion can be done in i) analog domain and ii) digital domain.
In analog domain using DAC the signal is converted into analog and then filtering is applied. Then the analog signal is converted back to digital using ADC. In digital domain all processing is done with signal in digital form. In the first method, the new sampling rate doesn't have any relationship with old sampling rate. But major disadvantage is the signal distortion. So the digital domain sampling rate conversion is preferred even then the new sampling rate depends on the old sampling rate.

14.What is the advantage & disadvantage of multirate signal processing?
Advantage: new sampling rate is selected which do not have any special relationship to old sampling rate.
Disadvantage: signal distortion introduced by D/A convertors.

15.What are the applications of sub band coding?
- Filter design.
- Synthesis method.
- Data compression in image signal processing.

16.What is speech synthesis? What are its applications?
A machine is developed which can accept as input a piece of English text and convert it to natural sounding speech.

Applications:
- Speech output from computers
- Reading machine for visually challenged.
- Accessing medical records stored in central computers.

17.Define speech coding. And its applications.
(OR) What do you mean by speech compression?

Speech coding is concerned with the developments of techniques which exploits the redundancy is the speech signal, in order to represent the less bits reduce the number of bits to present it.

Applications:

- Voice mail systems
- Cordless telephone channel
- Narrow band cellular radio
- Military communication

18.What are the classifications of speech sounds?

- Voiced sounds –vocal card vibrates produces quasi-periodic pulses.
- Fricative or unvoiced sounds – noise excitation.
- Plosive sounds-rapid release of pressure.

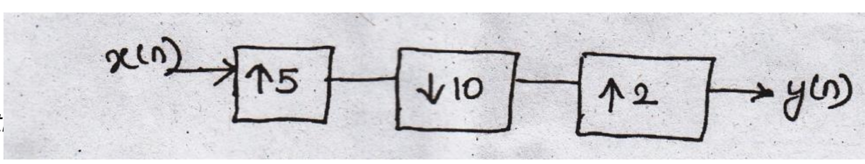
19.List the application of adaptive filtering?

- Noise cancellation
- Signal prediction
- Adaptive feedback cancellation
- Echo cancellation

PART-B

1. **Explain in detail the two basic operations in multi-rate signal processing.(16) (Apr/May 2015)**
2. **(i)Explain the concept of decimation and interpolation of discrete time signals.. (MAY 2012) (16 mark)**
   (ii)For the signal x(n), obtain the spectrum of down sampled signal x(Mn) and upsampled signal x(n/L).(may 2016)
   (iii) Explain sampling rate increase by an integer factor I and derive the input-output relationship in both time and frequency domains. (MAY 2013) (16 mark)
   (iv) Explain the concept of decimation by a factor D and interpolation by a factor I. with help of equation explain sampling rate conversion by a rational factor(I/D)(may 2017)
3. **Explain sampling rate conversion by a rational factor and derive input and output relation in both time and frequency domain. (DEC 2012) (8 mark)**
4. **Show that the upsampler and dowmsampler are time variant systems(Dec 2016)**
5. **A signal x(n) is given by x(n) = { 0,1,2,3,4,5,6,0,1,2,3 }**
   Obtain the decimated signal with a factor of 2.(ii)Obtain the interpolated signal with a factor of 2. (MAY 2013) (8 mark)
6. **Obtain the decimated signal y(n) by a factor 3 from the input signal x(n).(Dec 15)[Assume x(n) any values]**

7. **For the multirate system shown in figure develop an expression for the output y(n) as afunction of input x(n)**



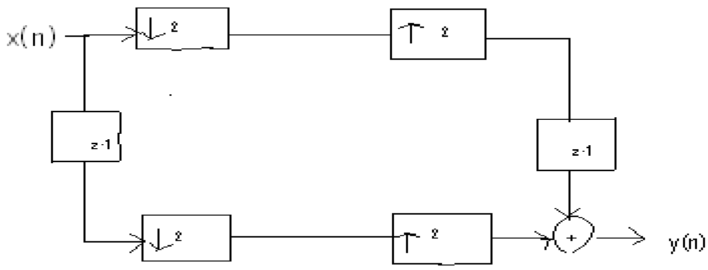
8. **Explain the efficient transversal structure for decimator and interpolator. (DEC 2011) (16 mark)**

9. **Draw the signal flow graph for IIR structure M to 1 decimator.(Dec15)**

10. **Draw the signal flow graph for 1 to L Interpolator.(Dec15)**

11. **Implement a two stage decimator for the following specifications.**
Sampling rate of the input signal = 20,000Hz,M=100, pass band = 0 to 40Hz,Transition band = 40 to 50 Hz, passband ripple = 0.01,stopband ripple =0.002.(Dec 2014,Dec 2015)
12. **explain in detail the multistage implementation of sampling rate conversion. (MAY 2012) (DEC 2012) (MAY 2013) (Dec 2016) (8 mark)**

13. **For the multirate system shown in figure. Find the relation between x(n) and y(n). (DEC 2011) (8 mark)**



14. **Explain the design of a narrow band filter using sampling rate conversion. (DEC 2012) (8 mark)**
15. **Explain sub and coding in detail. (MAY 2012) (8 mark)**
16. **Explain the application of sampling rate conversion in sub – band coding. (DEC 2012) (8 mark)**
17. **Discuss the procedure to implement digital filter bank using multirate signal processing. (Dec 10) (16 mark)**
18. **Explain how various sound effects can be generated with the help of DSP. (May 11) (8 mark)**
19. **Explain how DSP can be used for speech processing. (May 11) (8 mark)**
20. **Draw and explain the block diagram of subband coding system.(8)(Apr/May 2015)**
21. **Discuss about the musical sound processing.(8)(Apr/May 2015)**
22. **State the application of multirate signal processing (May 11) (8 mark)**
a. **(Explain subband coding , Vocoder ,)**
23. **Discuss in detail about any two applications of adaptive filtering with a suitable diagram.(May 2016).**
24. **Explain the operation of adaptive filter with suitable diagrams and equations.(May 2017)**

25. The frequency response of x(n) is shown in figure. If the input is passed through a down sampled by 2, find the frequency response of output and give your comment on alising.