UNIT-I - INTRODUCTION
PART – A

1. Define Signal.(Apr 2015)

A Signal is defined as any physical quantity that varies with time, space or any other independent variables. Ex: Speech signal, ECG

2. Define a system.(Apr 2015)

A System is a physical device (i.e., hardware) or algorithm (i.e., software) that performs an operation on the signal.

3. What are the steps involved in digital signal processing? (Apr 2015)

- Converting the analog signal to digital signal, this is performed by A/D Converter
- Processing Digital signal by digital system.
- Converting the digital signal to analog signal, this is performed by D/A converter.

4. Give some applications of DSP? (Apr 2015) ○ Speech processing – Speech compression & decompression for voice storage system ○ Communication – Elimination of noise by filtering and echo cancellation ○ Bio-Medical – Spectrum analysis of ECG, EEG etc.

5. Write the classifications of DT Signals.(Nov 2015)

- Energy & Power signals
- Periodic &A periodic signals
- Even & Odd signals.

6. What is an Energy and Power signal?(Nov 2015)

Energy signal:

A finite energy signal is periodic sequence, which has a finite energy but zero average power.

Power signal:

An Infinite energy signal with finite average power is called a power signal

7. What is Discrete Time Systems?(May 2014)

The function of discrete time systems is to process a given input sequence to generate output sequence. In practical discrete time systems, all signals are digital signals, and operations on such signals also lead to digital signals. Such discrete time systems are called digital filter.

8. Write the Various classifications of Discrete-Time systems.(May 2014) ○ Linear &Non linear system ○ Causal & Non Causal system ○ Stable &Un stable system ○ Static & Dynamic systems

9. Define Linear system(Nov 2014)

A system is said to be linear system if it satisfies Super position principle. Letus consider x1(n) & x2(n) be the two input sequences & y1(n) & y2(n) are the responses respectively,

$$T[ax1(n) + bx2(n)] = a\ y1(n) + by2(n)$$

10. Define Static & Dynamic systems(Nov 2014)

When the output of the system depends only upon the present input sample, then it iscalled static system, otherwise if the system depends pa t values of input then it is called dynamic system **11. Define causal system.**(Nov 2014)

When the output of the system depends only upon the present and past input sample, then it is called causal system, otherwise if the system depends on future values of input then it is called non-causal system

12. Define Shift-Invariant system.(Nov 2014)

If y(n) is the response to an inp t x(n), then the response to an input X(n) = x(n-n0) then y(n) = y(n-n0). When the system satisfies above condition then it is said to shift in variant, otherwise .it is Invariant.

13. Define impulse nd unit step signal. (Nov 2013)

Impulse signal (n):

The impulse signal is defined as a signal having unit magnitude at n = 0 and zero for other values of n.

u(n)= 1; n = 0

u(n) = 0; n 0

Unit step signal u(n):

The unit step signal is defined as a signal having unit magnitude or all values ofn 0 u(n)

= 1; n 0u(n)

=0; n>0

14. What are the basic elements used to construct the block diagram of discrete time system?

(May 2012)

The basic elements used to construct the block diagram of discrete time Systems are Adder, Constant multiplier &Unit delay element.

15. What is Truncation?(May 2012)

Truncation is the process of **reducing** the size of binary number by discarding all bits less significant than the least significant bit that is retained. (In the truncation of a binary number to b bits all the less significant bits beyond b^{th} are discarded)

16. Define sampling theorem.(Nov 2015)

A continuous time signal can be re resented in its samples and recovered back if the sampling frequency F_s2B. Here 'F_s' is the sampling frequency and 'B' is the maximum frequency present in the signal.

Since square root is nonlinear, the system is nonlinear.

As long as x(n) is bounded, its square root is bounded. Hence this system is stable.

17.What are the properties of convolution? (Nov 2012)

1. Commutative property x(n) * h(n) = h(n) * x(n)
2. Associative property [x(n) * h_1(n)]*h_2(n) = x(n)*[h_1(n) * h_2(n)]
3. Distributive property x(n) *[h1(n)+h2(n)] = [x(n)*h1(n)]+[x(n) * h2(n)

18. What is the difference between cross correlation and auto correlation?(Nov 2012)

1. Cross correlation operation is correlation of two different sequences, whereas auto correlation is correlation of a sequence with itself.
2. Auto correlation operationis an even function, whereas cross correlationis not an even function.

19.If x(n) = x(n+1) + x(n-2), is the system causal?(DEC 2016)

The system depends on future input.so the system is non-causal.

20. Define sampling theorem.

A continuous time signal can be represented in its samples and can be

Represented in its samples and can be recovered back when sampling frequency f_s is greater than or equal to the twice the highest frequency component of message signal.

PART – B

1. (i) Find the impulse response of a discrete time invariant system whose difference equation is given by $y(n) = y(n-1) + 0.5y(n-2) + x(n) + x(n-1)$.(12)**(Apr 2015)**
2. (i) A discrete time system is represented by the following difference equation in which x(n) is input and y(n) is output $y(n) = 3y(n-1) - nx(n) + 4x(n-1) + 2x(n+1)$; and n□0. Is the system linear? Shift invariant? Causal? In each case, justify your answer.(12)**(Apr 2015)**

3. (i) Check for linearity and time invariance $y(n) = x(2n) and\ y(n) = x(n)^2$.(8)**(Apr 2015)**

4. (i) What is energy and power signals? Determine whether the following signals are power or energy or neither of these two. (6) **(Apr 2015)**

$$(1)\ x(n) = \left({}^{1}/_{5}\right)^{n} u(n)$$
$$(2)\ x(n) = exp\left\{j\left(\frac{\pi n}{3}\right) + \left({}^{\pi}/_{7}\right)\right\}$$

7. (i) Check the causality and stability of the systems $y(n) = x(-n) + x(n-2) + x(2n-1)$(8)
(ii) Check the system for linearity and time invariance $y(n) = n(-1)x^2(n) + c$.(8) **(May2014)**
8. A discrete time system can be (i) Static or dynamic (ii) Linear or non-linear
(iii) Time invariant or time varying (iv) Stable or unstable
Examine the following system with respect to the properties above $y(n) = x(n) + nx(n+1)$
(16) **(May 2014)**

Theory

9.(i) Explain the following with respect to discrete time system: (6)**(Apr 2015)**
1. Causality
2. Stability
3. Dynamic system

(ii) What is meant by quantization and quantization error?(4) **(Apr 2015)**

10. (i) What is system? Explain the classification of systems.(5) **(Nov 2014)**
(ii) What is sampling? Explain the operation of sampling process.(7)
(iii) What is meant by spectral density? Explain. (4)

11. (i) Explain the properties of discrete time system.(4)**(Apr 2015)**

12. Distinguish the following with examples and formulae.
(i)energy vs power signal
(ii)time variant vs time invariant signal. (16)**(Dec 2016)**

UNIT-II - DISCRETE TIME SYSTEM ANALYSIS
PART – A

1. Define DTFT. (Apr 2015)

Let us consider the discrete time signal x(n).Its DTFT is denoted as X(w).It is given as $X(w) = x(n)e^{-jwn}$

2. State the condition for existence of DTFT? (Apr 2015)

The conditions are

If x(n)is absolutely summable then x(n)|<□

If x(n) is not absolutely summable then it should have finite energy for DTFT to exit.

3. List the properties of DTFT. (Apr 2015)

- Periodicity
- Linearity
- Time shift
- Frequency shift
- Scaling
- Differentiation in frequency domain
- Time reversal
- Convolution
- Multiplication in time domain
- Parseval's theorem

4. What is the DTFT of unit sample?(Apr 2015)

The DTFT of unit sample is 1 for all values of w.

5. Define DFT.(Nov 2015)

DFT is defined as X(k)= x(n)e^{-jwn}.

Here x(n) is the discretean updatestime sequence X(w) is the Fourier transform of x(n).

6. What is ROC in Z-Transform?(Nov 2015)

The values of z for which z – transform converges is called region of convergence (ROC). The ztransform has an infinite power series; hence it is necessary to mention the ROC along with z-transform. Linearity

Time Shifting

Frequency shift or Frequency translation

Time reversal

7. Define Zero padding.(May 2014)

The method of appending zero in the given sequence is called as Zero padding.

8. Define circularly even sequence.(May 2014)

A Sequence is said to be circularly even if it is symmetric about the point zero on the circle. x(N-n)=x(n),1<=n<=N-1.

9. Define circularly odd sequence.(Nov 2014)

A Sequence is said to be circularly odd if it is anti-symmetric about point x(0) on the circle

10. Define circularly folded sequences.(Nov 2014)

A circularly folded sequence is represented as x((-n)). It is obtained by plotting x(n) in clockwise direction along the circle.

11. State circular convolution.(May 2013)

This property states that multiplication of two DFT is equal to circular convolution of their sequence in time domain.

12. State Parseval's Theorem.(May 2013)

Consider the complex valued sequences x(n) and y(n).If x(n)y*(n)=1/N X(k)Y*(k) **13. Define Z transform.(Nov 2013)**

The Z transform of a discrete time signal x(n) is denoted by X(Z) and is givenby X(Z)= x(n)Z^{-n}.

14. Define ROC.(Nov 2013)

The value of Z for which the Z transform converged is called region of convergence. **15. Find Z transform of x(n)={1,2,3,4}(May 2012)**

$x(n) =$ {1,2,3,4} $X(z) =$
$x(n)z^{-n} =$ $1+2z^{-1}+3z^{-2}+4z^{-3}$.
$= 1+2/z+3/z^2+4/z^3$

16.Define time shifting property(May 2012)

The convolution property states that the convolution of two sequences in time domain is equivalent to multiplication of their Z transforms.

17.What z transform of (n-m)? (Nov 2015)

$Z[A\ (n-m)]=AZ^{-m}$ sin Z[(n)] =1

18. State initial value theorem. (Nov 2015)

If x(n) is causal sequence then its initial value is given by x(0)=lim X(z)

19. List the methods of obtaining inverse Z transform.(Nov 2012) Inverse z transform can be obtained by using Partial fraction expansion.
Contour integration
Power series expansion
Convolution

20. Obtain the inverse z transform of X(z)=1/z-a,|z|>|a|(Nov 2012)

Given X(z)=z-1/1-az-1
By time shifting property
$X(n)=a^n u(n-1)$

21.Find the system transfer function H(Z) if y(n) = x(n) + y(n-1) (Dec 2016)

$Y(Z) = X(Z) + Z^{-1}Y(Z)$

$Y(Z)[1-Z^{-1}] = X(Z)$

$H(Z)=Y(Z)/X(Z) = 1/[1-Z^{-1}] = Z/(Z-1)$

22.Explain the relationship between s-plane and z-plane. (Dec 2016)

Z is represented as $re^{j\omega}$ in polar form and relationship between Z plane and S plane is given as $Z=e^{ST}$ where $s= \sigma + j\ \Omega$.
$Z= e^{ST}$ (Relationship Between Z plane and S plane)
Z= e (σ + j Ω) T = e σ T . e j Ω T
Comparing Z value with the polar form we have. r = e σ T and ω = Ω T

1)Left side of s-plane is mapped inside the unit circle.
2) Right side of s-plane is mapped outside the unit circle.
3) jΩ axis is in s-plane is mapped on the unit circle.

PART – B

Problems

1. (i) find the Z transform of $x(n) = n^2 u(n)$

(ii) find the inverse Z – transform of $X(Z) = \frac{z}{3z^2-4z+1}$ for Region of convergence
(1) $|Z| > 1$ (2) $|Z| < 1/3$ (3) $\frac{1}{3} < |Z| < 1$

(16) **(Apr 2015)**

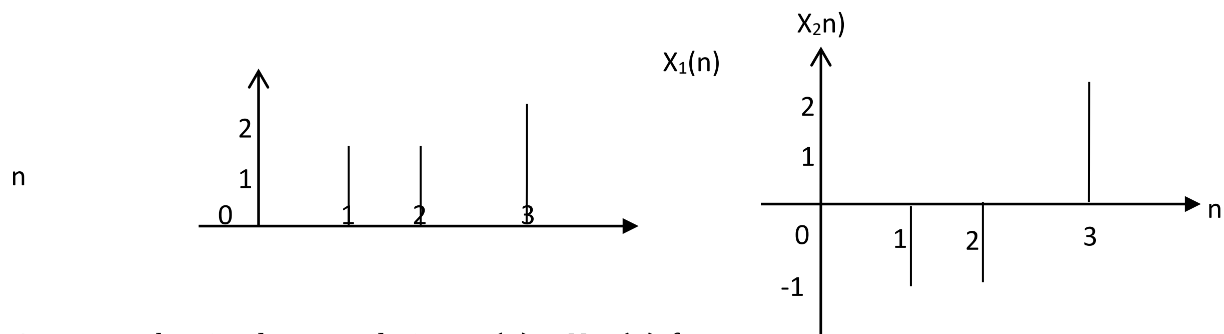
2.(i) Convolute the following two sequences $x_1(n) = \{0,1,4,-2\}$ and $x_2(n) = \{1,2,2,2\}$.
(8)**(Apr 2015)**

(ii) find the frequency response of the LT1 system governed by the equation $y(n) = a_1 y(n-1) - a_2 y(n-2) - x(n)$. (8)**(Apr 2015)**

3.(i) Using scaling property, determine the z-transform of the sequence $x(n) = a^n Cos\omega_0 n$

(ii) Find x(n) by convolution for $X(Z) = \frac{1}{(1-0.5z^{-1})(1+0.25z^{-1})}$ **(Apr 2015)**

4.(i) Given the sequence $x_1(n)$ and $x_2(n)$ as shown below (8)**(Apr 2015)**



Computer the circular convolution $x_1(n) * N x_2(n)$ for N =4. What is the value of N for which this yields the same result or linear convolution?

5.(i) Find the Z transform and its ROC of $x(n) = \left(\frac{1}{2}\right)^{|n|} + \left(-\frac{1}{2}\right)^{|n|}$ (10)

ii) Find the linear convolution of x(n) = {1, 2, 3, 4, 5,6, 7,} with h(n)={2, 4, 6, 8} (6)

6.(i) What is frequency response? Explain its Properties. (6) **(Nov 2014)**

ii) Find the inverse z-trasnform of $X(z) = \frac{4Z}{(Z+1)^2(Z+3)}$ for all possible ROCs. (10)

7. (i) Find the Z – transform and ROC of $x(n) = r^n \cos(n\theta)\, u(n)$ (8) **(May 2014)**

(ii) Find Inverse Z – Transform of $X(z) = z/[3z^2 - 4z + 1]$, ROC $|z| > 1$ (8)

8.(i) Determine the DTFT of the given sequence $x(n) = a^n(u(n) - u(n-8))$, $|a| < 1$ (8)

(May 2014)

9. (a) Compute the following if: x_1 = [-1,-1,-1,2]; x_2 = [-2,-1,-1,-2]

(i) Linear and circular convolution of a sequence

(ii) x_1 ; x_2 subject to addition and multiplication.

10.Evaluate the following. **(Dec 2016)**

(i) The impulse response h(n) for y(n)=x(n)+2x(n-1)-4x(n-2)+x(n-3) (ii) The ROC of a finite duration signal x(n)={2,-1,-2,-3,0,-1}
(iii) Inverse Z-Transform for X(z)= 1/(z-1.5); ROC :|Z|>1/4.

11. What is the need for frequency response analysis? Determine the frequency response and plot the magnitude response and phase response for the system.

   Y(n)=2x(n)+x(n-1)+1y(n-2) (**Dec 2016)**

   Theory

12. How is the DFT obtained from discrete time Fourier Transform? Why? (8)**(Apr 2015)**
13. Prove the linearity and frequency shifting theorem of the DTFT. (8)**(May 2014)**

UNIT III

DISCRETE FOURIER TRANSFORM & COMPUTATION

1.Define DFT and discrete time sequence.(Arp 2015).

The DFT and IDFT pair is given by

$$DFT \Rightarrow X(k) = \sum_{n=0}^{N-1} x(n)e^{-j\left(\frac{2\pi}{N}\right)nk} \qquad k = 0,1..(N-1)$$

$$IDFT \Rightarrow x(n) = \sum_{k=0}^{N-1} X(k)e^{+j\left(\frac{2\pi}{N}\right)nk} \qquad n = 0,1..(N-1)$$

2.Define Twiddle factor.(May 2014)(Arp 2015).

To simplify the notation the complex valued phase factor or twiddle factor $W^K{}_N = e^{-j2\square k/N}$

$$DFT \Rightarrow X(k) = \sum_{n=0}^{N-1} x(n)e^{-j\left(\frac{2\pi}{N}\right)nk} \qquad k = 0,1..(N-1)$$

Here, W represents a complex number 1 $\llcorner$-2π. Hence Phase of W is $\llcorner$-2π. Therefore, when a number is multiplied by W, only its phase changes by $\llcorner$-2π but magnitude remains same.

3.What are the differences and similarities between DIF and DIT diagram? (Arp 2015).

Differences:

a. In DIT the time domain sequence is decimated, whereas in DIF the frequency domain sequence is decimated.
b. For DIT, the input is bit reversal while the output is of natural order whereas for DIF, the input is in natural order while the output is bit reversal.
c. The DIF butterfly is slightly different from the DIT butterfly, the difference being that the twiddle factor multiplication takes place after the add /sub operation in DIF.

Similarities:

a. Both algorithms require same number of options to compute the DFT. Both algorithms can be done in place.
b. Both require bit reversal at some place during computation.

4.Mention any two application of DFT.(Arp 2015).

i. Linear filtering ii. Correlation
iii. Spectrum Analyses

5.List any four properties of DFT.(Nov 2014)

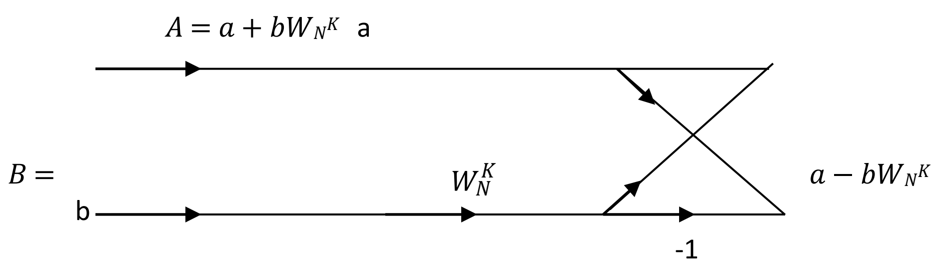
i. Periodicity ii. Linearity
iii. Time reverse sequence
iv. Circular time shifting of a sequence

6.Explain Fourier series.(Nov 2014)

Consider a discrete time fourier sequence x(n) with a period of N samples so that x(n) = x(n-N), the discrete Fourier series of the sequence x(n) is defined as

$$X(k) = \sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N}$$

7.Draw the basic butterfly diagram of DIT radix – 2 FFT(May 2014)



8. What is meant by radix 2 FFT?

The FFT algorithm is most efficient in calculating N-point DFT. If the no. of o/p points N can be expressed as a power of 2 that is 2^M, where M is an integer, then this algorithm is known as radix 2FFT algorithm.

9. Why the result of circular and linear convolution is not same?

In linear convolution the length of the convoluted sequence is equal to (M+N-1) length of two sequences minus one, sozero padding is not necessary in linear convolution. But in circular convolutionthe length of convoluted sequence is equal to M if (M >N)Due to it the samples get aliased in circular convolution and recovering is a tough process.

10. What is zero padding? What are its uses?(Dec 2016)

Let the sequence x(n) has a length, L. If we want to find the N-point DFT if N>L of the sequence x(n), we have to add (N – 1) zeros to the sequence x(n). This is known zero padding. The uses of padding a sequence with zeros are

a. We can get “better display” of the frequency spectrum.
b. With zero padding, the DFT can be used for linear filtering.

11. Why FFT is needed?

The direct evaluation of DFT using the formula

$$X(k) = \sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N}$$

requires N^2 complex multiplication and N(N – 1) complex additions. Thus for reasonably large values N direct evaluation of the DFT requires an un ordinate amount of computation. By using FFT algorithm the no. of computations can be reduced to (N/2)log2N complex multiplication and Nlog2N complex addition.

12. What is FFT?

The FFT is an algorithm used to compute the DFT. It makes use of the symmetry and periodicity properties of twiddle factor W_N^k to effectively reduce the DFT computation time. It is based on the fundamental principle of decomposing the computation of DFT of a sequence of length N into successively smaller DFT. The FFT algorithm increase speed when compared with direct computation of the DFT.

13. Define circular convolution?

Let $x_1(n)$ and $x_2(n)$ are finite duration sequence both of length N and DFTs $X_1(k)$ and $X_2(k)$. If $X_3(k) = X_1(k)\, X_2(k)$, then the sequence $x_3(n)$ can be obtained by circular convolution, defined as

$x_3(n) = \sum_{m=0}^{N-1} x_1(m) x_2((n-m))_N = x_1(n) \circledast x_2(n) = X_1(k)\, X_2(k)$

14. State Parseval's relation of DFT.

Parseval's relation is given by

If $x(n) \overset{DFT}{\longleftrightarrow} X(k)$

then

$$\sum_{n=0}^{N-1} |x(n)|^2 = \frac{1}{N} \sum_{k=0}^{N-1} |X(k)|^2$$

15. How to obtain circular convolution from Linear convolution?

Consider two finite duration seq. x(n) and h(n) of M samples and N samples respectively. The linear convolution of these two sequences produces an output sequence of duration M+N-1 samples, whereas, the circular convolution of x(n) and h(n) gives N samples where N=Max(N,M). In order to obtain the no. of samples in circular convolution equal to linear convolution, both x(n) and h(n) must be appended with appropriate no. of zero valued samples. In other words by increasing the length of the sequences x(n) and h(n) to (N+M1) points and then circularly convoluting the resulting sequence we obtain the same results as that of linear.

16. Distinguish between DFT & DTFT.

DFT	DTFT
Obtained by performing sampling operation in both the time and frequency domain.	Sampling is performed only in time domain
Discrete frequency spectrum	Continuous function of □.
$X(K) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi nk/N}$ $k = 0,1, \dots .(N-1)$	$X(e^{j\omega}) = \sum_{n=-\infty}^{\infty} x(n) e^{-j\omega n}$

17. Establish the relationship between DFT and Z-transformer.

The Z-transform of a sequence x(n) is

$$X(z) = \sum_{n=-\infty}^{\infty} x(n) Z^n$$

With ROC that includes the unit circle.

If X(z) is sampled at N equally spaced point on the unit circle $Z_k = e^{j2□k/N}$,

k = 0, 1,……… (N – 1)

$$X(k) = X(z)\big|z = e^{i2\pi k}/N\big| \qquad K = 0,1, \dots .(N-1)$$

$$= \sum_{n=0}^{N-1} x(n) e^{-j2n\pi k/N}$$

$$X(z) = \sum_{n=0}^{\infty} x(n) z^{-n}$$

$$= \sum_{n=0}^{\infty} \left[\frac{1}{N} \sum_{k=0}^{n-1} X(k) e^{-j2n\pi k/N} \right] z^{-n}$$

$$= \frac{1}{N} \sum_{K=0}^{N-1} X(k) \sum_{n=0}^{N-1} \left(e^{-j2n\pi k/N} Z^{-1} \right)^n$$

$$X(z) = \frac{1}{N} \sum_{K=0}^{N-1} \frac{X(k)}{1 - e^{-j2n\pi k/N} z^{-1}}$$

18. State circular frequency shift property of DFT.

The Circular frequency shift property of DFT says that if a discrete time signal is multiplied by $e^{\frac{j2\pi mn}{N}}$ its DFT is circularly m units.

$$DFT\left[x(n) e^{\frac{j2\pi nl}{N}} = X((K-l))_N \right]$$

19. Compare DIT and DIF in radix-2 FFT.

DIT radix-2 FFT	DIF radix-2 FFT
The time domain sequence is decimated	The frequency domain sequence is decimated.
The input should be in bit reversed order, the output will be in normal order	The input should be in normal order, the output will be in bit reversed orders.
In each stage of computations, the phase factors are multiplied before add and subtract operations.	In eachstage of computations, the phase factors are multipled after add and subtract operations.
Total number of arithmetic operations are $N\log_2 N$ complex additions and $(N/2)\log_2 N$ complex multiplications	Total number of arithmetic operations are $N\log_2 N$ complex additions and $(N/2)\log_2 N$ complex multiplications.

PART B

Problems

1. Determine the DFT of the sequence $x(n) = \begin{cases} \frac{1}{4}, & for\ 0 \le n \le 2 \\ 0, & Otherwise \end{cases}$ (8)**(Arp 2015).**

2. i) Given x(n) = n+1,and N = 8,find X(K) using DIT- FFT algorithm.(8) ii) Use 4-point inverse FFT for the DFT result {6,-2+j2,-2,-2-j2} and determine the input sequence. (8)**(Apr 2015)**

3. Determine and plot the magnitude and phase response of three point moving average system given by y(n) = 1/3[x(n+1)+x(n)+x(n-1)](8) **(Apr 2015)**

4. i)Obtain eight point DFT of the input sequence x(n) = {1,1,1,1,1,1,1,1} using decimation in frequency fast fourier transform algorithm. (10)**(Arp 2015).**

5. Find the 4-point inverse DFT of X(K) = {10, -2+j2, -2, -2-j2}.(8) **(Nov 2014)**

   ii) Obtain the 8- point DFT of the sequence x(n) = {1,1,1,1}(8) **(Nov 2014)**

6. An 8-point sequence is given by x(n) = {2,2,2,2,1,1,1,1} compute DFT of x(n) using radix 2 DIT FFT.(16)**(May 2014)**
7. i)Determine 8 point DFT of the sequence x(n) ={1,1,1,1,1,1,0,0}.(8) ii)Find circular convolution of the sequence using concentric circle method x1 = {1,1,2,1} and x2 ={1,2,3,4}.(8) **(May 2014)**
8. Describe the need for Bit reversal and Butterfly structure. For a sequence x(n)= {4,3,2,1,-1,2,3,4} obtain the 8pt FFT computation using DIT method**.(Dec 2016)**

Theory

9. Draw the flow graph of an 8- point DFT FFT algorithm and explain. (8) **(Arp 2015).**
10. Prove the periodicity and time reversal properties of Discrete Time Fourier Transform.(8)**(Arp 2015).**
11. How is the FFT algorithm applied to determine inverse discrete fourier transform(6**).(Apr 2015)**
12. State and prove convolution property of DFT.(8)**(Nov 2014)**
13. Derive decimation in frequency, radix-2,FFT algorithm for evaluating DFT.(8)

UNIT-IV - DESIGN OF DIGITAL FILTER PART-A

1. **What are the special features of FIR filter?(Apr 2015)**
   - FIR filters have exact linear phase
   - FIR filters are always stable
   - FIR filters can be realized in both recursive and non-recursive structure
   - Filters with any arbitrary magnitude response can be tackled using FIR sequence.

2. **What is meant by pre-warping? (Apr 2015)**

The effect of the non-linear compression at high frequencies can be compensated. When the desired magnitude response is piece-wise constant over frequency, this compression can be compensated by introducing a suitable pre-scaling or pre-warping the critical frequencies by using the formula

$$\Omega = \frac{2}{T} tan \frac{\omega}{2}$$

3. What are the significance of Chebyshev filter? (Apr 2015)

1. The magnitude response of Chebyshev filter exhibits ripple factor either in passband or in stopband according to type.
2. Poles of the Chebyshev filter lie on an ellipse.

4. What is warping effect? What is its effect on magnitude and phase response? (Apr 2015)

The relation between the analog and digital frequencies in bilinear transformation is given by

$$\Omega = \frac{2}{T} tan \frac{\omega}{2}$$

For smaller vales of w there exist where relationship between □ and □. But for large values of W the relationship is non-linear. This non-linear introduces distortion in the frequencies. This is known as warping effect. The effect of compression of the magnitude and phase response at high frequency.

5. **What is the reason that FIR filter is always stable?(Nov 2014)** FIR filter is always stable because all its poles are at the origin.

6. **State the condition for a digital filter to be causal and stable**.**(Nov 2014)** A digital filter is causal if its impulse response h(n) = 0 for n < 0.

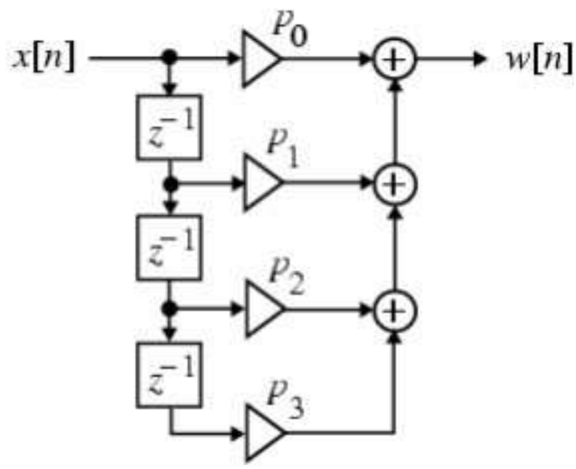
A digital filter is stable if its impulse response is absolutely summable

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

7. Mention the methods to convert analog filters into digital filters. (May 2013)

a. Impulse invariance method
b. Bilinear transformation method

8. For a general IIR filter draw the direct form transposed structure.(May 2013)



9. Write the relationship between the analog and digital frequencies is converting an analog filter to digital filer using impulse invariant method. (Nov 2013)

$$\frac{1}{S-P_k} \rightarrow \frac{1}{1-e^{P_K T}z^{-1}}$$

$$i.e.\, H(z) = \sum_{K=1}^{N} \frac{C_k}{1-e^{P_K T}z^{-1}}$$

10. What are the disadvantages of FIR filter? (Nov 2013)

- For the same filter specification, the order of FIR filter design can be as high as 5 to 10 times that of an IIR design
- Large storage requirement needed
- Powerful computational facilities required for the implementation

11. What are the properties of FIR filter? (May 2012)

a. FIR filter is always stable
b. A realizable filter can always be obtained
c. FIR filter has a linear phase response.

12. What are the advantages of Kaiser window? (May2012)

a. It provides flexibility for the designer to select the side lobe level and N.
b. It has the attractive property that the side lobe level can be varied continuously from the low value in the Blackman window to the high value is the rectangular window.

13. Compare Hamming and Kaiser window.(Nov2015)

Hamming	Kaiser
The main lobe width is equal to $\frac{8\pi}{N}$ and the peak side lobe level is -41 db	The main level width the peak lobe level can be varied by varying the parameters. □ and N
The low pass FIR filter designed will have first side lobe peak of -53Db	The side lobe peak can be varied by varying the parameter □

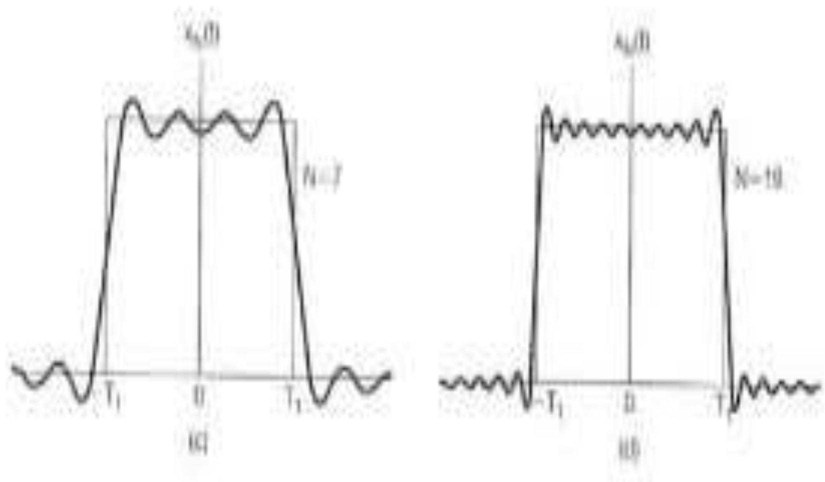
14. What is the necessary and sufficient condition for the linear phase characteristics of a FIR filters? (Nov 2015)

The conditions are

i. Symmetric condition h(n) = h (N-1-n)
ii. Antisymmetric condition h(n)= - h (N-1-n)

15. What is Gibbs phenomenon (or) oscillations? (Nov2012)

One possible way of finding an FIR filter that approximates. $H(e^{jw})$ would be to truncate the infinite Fourier series at n = I $\left(\frac{N-1}{2}\right)$. Abrupt truncation of the series will lead to oscillation both in passband and in stop band. This phenomenon is known as Gibbs phenomenon.



16. What are the desirable characteristics of the window? (Nov2012)

a. The central lobe of the frequency response of the window should contain most of the energy and should be narrow.
b. The highest side lobe level of the frequency response should be small.
c. The side lobes of the frequency response should decrease in energy rapidly as ω tends to □.

17. What are the conditions to be satisfied for constant group and phase delay?(May 2013)

For linear phase FIR filter to have both constant group delay and constant phase delay.

□(□) = -□w -□□□□□

For satisfying above condition

i.e, the impulse response must be symmetrical about $n = \frac{N-1}{2}$

If only constant group delay is desired, then

□(□) = □ - □w

For satisfying the above condition h(n)
= -h(N-1-n)

i.e, the impulse response is anti-symmetrical about
$n = \frac{N-1}{2}$

18. Write the properties of Butterworth filter. ?(May 2013)

i. The Butterworth filters are all pole design.
ii. At the cutoff frequency□2, the magnitude of normalized Butterworth filter is 1□2
iii. The filter order N, completely specifies the filter and as the value of N increases the magnitude response approaches the ideal response.
iv. The magnitude is maximally flat at the origin and monotonically decreasing with increasing □.

19. Write the properties of Chebyshev type-1 filters.(Dec 2016)

i. The magnitude response is equi-ripple in the passband and monotonic in the stopband.
ii. The Chebyshev type-1 filters are all pole design.
iii. The normalized magnitude function has a value of $\frac{1}{\sqrt{1+\varepsilon^2}}$ at the cutoff frequency □e.
iv. The magnitude response approaches the ideal response as the value of N increases.

20. Compare the Butterworth and Chebyshev Type-1 filters.(Dec 2016)

Butterworth	Chebyshev Type-1

The poles lie on a circle in s-plane.	The poles lie on an ellipse circle in s-plane.
The magnitude response is maximally flat at the origin and monotically decreasing function of □	The magnitude response is equi-ripple in band pass and monotonically decreasing in the stopband.

PART B

Problems

1. i) A low pass filter is to be designed with the following desired frequency response.

$$H_d(e^{j\omega}) = \begin{cases} e^{-j2\omega}, & \frac{-\pi}{4} \le |\omega| \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} < |\omega| \le \pi \end{cases}$$

Determine the filter coefficients $h_d(n)$ if the window function is defined as

$$\omega(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & Otherwise \end{cases}$$ (16) **(Apr 2015).**

2. Determine H(z) for a butter worth filter satisfying the following constraints.

With T =1s.Apply impulse invariant transformation.

$$0.75 \le |H(e^{j\omega})| \le 1 \quad 0 \le \omega \le \pi$$

$$|H(e^{j\omega})| \le 0.2 \quad 0.5\pi \le \omega \le \pi$$ (16) **(Apr 2015)**

3. i)Determine the cascade and parallel realization for the system transfer function

$$H(z) = \frac{3(2z^2+5z+4)}{(2z+1)(z+2)}$$ (10)

ii) What is Hamming window function?Obtain its frequency domain characteristics.(6) **(Apr 2015)**

4. Design a digital butter worth filter that satisfies the following constraints using bilinear transformation.Assume T = 1s.

$$0.9 \le |H(e^{j\omega})| \le 1 \quad 0 \le \omega \le \frac{\pi}{2}$$

$$|H(e^{j\omega})| \le 0.2 \quad 3\pi/2 \le \omega \le \pi$$ (16)**(Apr 2015)**

5. i) Implement the following system function using cascade structure. (6)

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

ii) Design a low pass filter for the following specifications using rectangular window function: (10)**(Nov 2014)**
cut-off frequency 500Hz Sampling frequency = 200 Hz
Order of the filter = 10

6. i)Convert the following analog transfer function into digital using impulse invariant technique with sampling period T = 1 sec.

$$H(Z) = \frac{s+1}{(s+3)(s+5)}$$ (10)**(Nov 2014)**

7. Using a rectangular window technique design a LPF with pass band gain of unity,cutoff frequency of 1000Hz and working sampling frequency of 5kHz.The length of impulse is 7.(16) **(May 2014)**
8. Design a Chebyshev filter for the following specification using bilinear transformation. (16)**(May 2014)**

$$0.8 \le |H(e^{j\omega})| \le 1 \quad 0 \le \omega \le 0.2\pi$$

$$|H(e^{j\omega})| \le 0.2 \quad 0.6\pi \le \omega \le \pi$$

9. A difference equation describing a filter is given by y(n) – 2y(n-1) + y(n+2) = x(n)+ 1/2x(n) obtain direct form II structure.(10)

10. Obtain the system function of the digital filter if the analog filter is $H_a(s) = 1/[(s+0.2)^2 +2]$. Using the impulse invariance method and the bilinear Transformation method obtain the digital filter. (16) **(Dec2016)**

11. A difference equation describing a filter is given by
    y(n) -2y(n-1) + y(n+2) = x(n)+1/2x(n-1) obtain direct for II structure.

12. Obtain the system function of digital filter if the analog filter is $H_a(s) =1/[(s+0.2)^2 + 2]$. Using the impulse invariance method and the Bilinear Transformation method obtain the digital filter.**(16)(Dec2016)**

13. (iii) Compute numerically the effect of hamming windows and design the filter if
    Cut off frequency = 100 Hz
    Sampling frequency = 1000 Hz
    Order of filter = 2
    Filter length required = 5 (16)

Theory

14. Explain what is meant by warping? (6)
15. Writebriefly on any TWO of the following (8+8)**(Dec2016)**
    (i) Comparison of Butterworth and Chebyshev Filter.
    (ii) Elaborate one application of digital signal processing with a DS processor.
16. (a) (i) Explain the role of windowing to realize a FIR filter**(Dec2016)**

(ii) Compare and explain on the choice and type of windows selection for signal analysis.

17. (a) Write briefly on any TWO of the following**(Dec2016)**

    (i) Comparison of Butterworth and Chebyshev Filter

UNIT-V - DIGITAL SIGNAL PROCESSORS PART-A

1. **What are the classifications of DSP?(Apr 2014)**
   a. General purpose DSP
   b. Special purpose DSP

2. **Write short notes on general purpose DSP processors (June 2016)**

General-purpose digital signal processors are basically high speed microprocessors with hard ware architecture and instruction set optimized for DSP operations. These processors make extensive use of parallelism, Harvard architecture, pipelining and dedicated hardware whenever possible to perform time consuming operations

Give some examples of floating pt DSPs

TMS320C3X, TMS320C67X, ADSP – 21XXX

3. What are the factors that influence solution of DSP$_S$? **(Apr 2014)**

   a. Architectural features
   b. Execution speed
   c. Type of arithmetic
   d. Word length

4. What are the applications of PDSP$_C$?
   Digital cell phones, automated inspection, voicemail, motor control, video conference, noise cancellation, medical imaging, speech synthesis, satellite communication.

5. What is pipelining?
   Pipelining a processor means breaking down its instruction into a series of discrete pipeling stage which can be completed is sequence by specialized hardware.

6. What is pipeline depth?
   e. no. of pipeline stages is referred to as the pipeline depth.

7. What is the pipeline depth of TMS320C50 and TMS32054X?
   TMS 320C50-4
   TMS 320C54X – 6

8. What are the different stages in pipelining?**(Apr 2014)**

   The fetch phase
   The decode phase
   Memory read phase
   Execute phase
9. What are the different buses of TMS32OC5X?**(Apr 2014)**

   Program bus
   Program address bus
   Data read bus
   Data read address bus

10. List the various registers used with ARAU**(Nov 2013)**
    Eight auxiliary registers (AR0-AR7)
    Auxiliary register pointer(ARP)
    Unsigned 16-bit ALU

11. What is the function of parallel log unit?
    The parallel logic unit is a second logic unit, that execute logic operations on data without affecting the contents of accumulator.

12. What are the general purpose I/O puns?
    Branch control input $(\overline{BIO})$
    External flag (XF)

13. What are the arithmetic instructions of C5X?
    ADD, ADDB, ADDC, SUB, SUBB, MPYU, MPY

14. What are the logical instruction of C5X?
    AND, ANDB, OR, ORB, XOR, XORB

15. What are the shift instructions?

ROR, ROL, ROLB, RORB BSAR

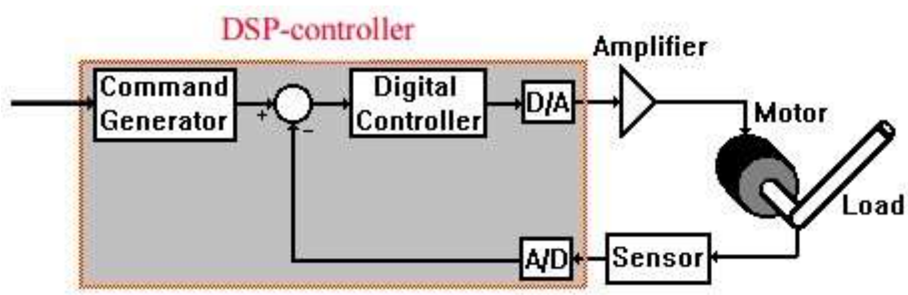
16. What are the load / store instructions?**(Nov 2014)**

LACB, LACC, LACL, LAMM, LAR, SACB, SACH, SACL, SAR, SAMM

17. What are the addressing modes of TMS320CS4x processors?**(Nov 2014)** The TMS320C54x processors supports the following seven addressing modes.
    20. Immediate addressing
    21. Absolute Addressing
    22. Accumulator addressing
    23. Direct addressing
    24. Indirect addressing
    25. Memory – mapped register addressing
    26. Stack addressing
18. Write a short note on MAC unit in DSPs.**(May 2013)**
    The MAC unit is DSPs is capable of performing multiply-add operations involved in convolution and correlation. A typical MAC unit consists of a multiplier, a temporary register, a product register, an adder and an accumulator. Initially the product register and accumulator are cleared and then MAC instruction is executed, a number of times required to compute one data of convolution. The execution of MAC instruction will add the content of P-register to accumulator and multiply two memory data and store the product in P-register.
19. How is fast data access achieved in digital signal processors?**(May 2013)**
    In digital signal processors, the fast data access is achieved by high-band width memory architecture like modified Harvard architecture, specialized addressing modes like circular and bit reversed addressing and DMA.

22.What is the advantage of Harvard architecture in a DS processor? (Dec 2015)

In Harvard architecture, memory of data and memory of instruction are separated. Its advantages includes, faster execution time:it allow concurrent access of data and instruction 23.How is a DS processor applicable for motor control applications (Dec 2015)



24. What are the merits and demerits of VLIW architecture?(May 2016)

Advantages:Increased performance, Better compiler targets,Potentially easier to program,

Disadvantages:Increased memory use,High program memory bandwidth requirements,High power consumption,Misleading MIPS ratings

25. What are the factors that influence the selection of DSP processor for an application (May 2016)

The right DSP processor for a job depends heavily on the application. One processor may perform well for some applications, but be a poor choice for others. With this in mind, one can consider a number of features that vary from one DSP to another in selecting a processor. These features are: Ease of Development,Multiprocessor Support,Power Consumption and Management, Cost,Memory Organization.