Doc. No.: DCE/0/15
Revision :00

# Lecture Plan -1

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP Unit:-I

| S. No. | Topic :-VARIOUS CLASSIFICATIONS OF SIGNALS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>The various classification of signals in terms of its properties as varying discretely or continuously with time or repeating itself after specific interval of time or not. | 5min |
| 2. | **Division of the Topic**<br>-DISCRETE-CONTINOUS SIGNAL<br>-PERIODIC-APERIODIC SIGNAL<br>-ENERGY-POWER SIGNALS | 35 min |
| 3. | **Conclusion**<br>A signal carries some useful information. Discrete time signal varies randomly with time, periodic repeats periodically. | 5min |
| 4 | **Question / Answer**<br>Q1-How will you classify whether the signal is energy ?<br>A1-if energy calculated is finite & power being zero then energy signal.<br>Q2-What is difference bw deterministic & non deterministic signal?<br>A2-deterministic signal is random in nature. | 5min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLSKHY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-2

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP Unit:-I

| S. No. | Topic :-SIGNAL REPRESENTATION IN TIME & FREQUENCY DOMAIN | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>TIME & FREQUENCY DOMAIN REPRESENTATION of the signal is one in which the variation of signal with time & frequency are drawn respectively. | 5min |
| 2. | **Division of the Topic**<br>-TIME DOMAIN: VARIES DIRECTLY WITH TIME<br>-FREQUENCY DOMAIN: FOURIER SERIES ,FOURIER TRANSFORM | 30min |
| 3. | **Conclusion**<br>**Fourier series** is frequency domain representation for periodic signals.<br>**Fourier transform**: aperiodic signals | 5min |
| 4 | **Question/answer**<br>Q1-What is difference bw FOURIER series & FOURIER transform?<br>A1- FOURIER series is drawn for periodic signal where as FOURIER transform is for non periodic signal.<br>Q2-how many types of FOURIER series are there?<br>A2-trignometric & complex | 5min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ , OPPENHELM & WILLSKHY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-3

Semester:-VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-I

| S. No. | Topic :-FOURIER TRANSFORM & PROPERTIES OF FOURIER TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Introduction to fourier transform.. | 5min |
| 2 | **Division of the Topic**<br>-Fourier transform<br>-properties of fourier transform | 30min |
| 3. | **Conclusion**<br>Fourier transform & its properties are useful in many applications. | 10 min |
| 4 | **Question / Answer**<br>Q1-will you get the Fourier series/Fourier transform of full wave rectifier?<br>A1 fourier series. | 5 min |

Assignment to be given:-OBTAIN THE FOURIER SERIES FOR FULL WAVE RECTIFIER.

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLSKHY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-4

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-I

| S. No. | Topic :-FOURIER TRANSFORM & PROPERTIES OF FOURIER TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Introduction to fourier transform.. | 5min |
| 2 | **Division of the Topic**<br>-Fourier transform<br>-properties of fourier transform | 30min |
| 3. | **Conclusion**<br>Fourier transform & its properties are useful in many applications. | 10 min |
| 4 | **Question / Answer**<br>Q1-will you get the Fourier series/Fourier transform of full wave rectifier?<br>A1 fourier series. | 5 min |

Assignment to be given:-OBTAIN THE FOURIER SERIES FOR FULL WAVE RECTIFIER.

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLSKHY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan -5

Semester:- VI Class:-ECS Course Code:-EC-614-F Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-I

| S. No. | Topic :-SYSTEM PROPERTIES | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>VARIOUS PROPERTIES OF SYSTEMS ON THE BASIS OF ITS VARIATION WITH TIME, DISTRIBUTION PROPERTY. | 5 min |
| 2 | **Division of the Topic**<br>Time invariance<br>Distributive ,cumulative & associative property | 30min |
| 3. | **Conclusion**<br>A system which does not varies with time & obeys superposition principal is LTI sys & being used in most of the applications. | 10 min |
| 4 | **Question / Answer**<br>Check for LTI property:<br>Q1-Y(n)= x(n^2}<br>A1-not linear & time invariant.<br>Q2-Y(n)=a* x(n)<br>A2-linear & time invariant | 5 min |

**Assignment to be given**:-NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-6

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-II

| S. No. | Topic :-SYSTEM CLASSIFICATION | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Various types of systems are studied on the basis of its properties it follows as linearity, casuality & stability. | 5 min |
| 2 | **Division of the Topic**<br>-Types are: CASUAL- ANTICAUSAL<br>-LINEAR- NON LINEAR<br>-STABLE -UNSTABLE | 30min |
| 3. | **Conclusion**<br>CASUAL- having present & past i/p; STABLE- BIBO STABILITY<br>LINEAR- superposition principle is followed. | 10 min |
| 4 | **Question / Answer**<br>Q1-Check whether the signals are causal?<br>A1-IF system has only past & present i/ps then it is casual.<br>Q2-what are the properties of linear & causal system?<br>A2-if it follows superposition principal & BIBO stability condition. | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ,OPPENHELM & WILLSKY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-7

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-II

| S. No. | Topic :-SYSTEM CLASSIFICATION | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Various types of systems are studied on the basis of its properties it follows as linearity, casuality & stability. | 5 min |
| 2 | **Division of the Topic**<br>-Types are: CASUAL- ANTICAUSAL<br>-LINEAR- NON LINEAR<br>-STABLE -UNSTABLE | 30min |
| 3. | **Conclusion**<br>CASUAL- having present & past i/p; STABLE- BIBO STABILITY<br>LINEAR- superposition principle is followed. | 10 min |
| 4 | **Question / Answer**<br>Q1-Check whether the signals are causal?<br>A1-IF system has only past & present i/ps then it is casual.<br>Q2-what are the properties of linear & causal system?<br>A2-if it follows superposition principal & BIBO stability condition. | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ,OPPENHELM & WILLSKY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-8

Semester:- VI Class:-ECS Course Code:-EC-614-F Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-II

| S. No. | Topic :-ENERGY & POWER THEORMS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>PARSEVAL'S ENERGY & POWER THEORMS for calculation of energy & power for signals. | 5min |
| 2 | **Division of the Topic**<br>-PARSEVAL'S ENERGY THEORM<br>-PARSEVAL'S POWER THEORM | 30min |
| 3. | **Conclusion**<br>Energy signal is one having P=0,E= FINITE is the main condition and for POWER SIGNAL: P=FINITE,E=INFINITE | 10 min |
| 4 | **Question / Answer**<br>Q1-Obtain the energy of periodic signal?<br>A1-it is finite<br>Q2-will you get the Fourier series/Fourier transform of full wave rectifier?<br>A2-fourier series. | 5 min |

Assignment to be given:-OBTAIN THE FOURIER SERIES FOR FULL WAVE RECTIFIER.

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLSKHY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan9

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-II

| S. No. | Topic :-LINEAR TIME INVARIENT SYSTEM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>A Linear time invariant system is defined as the system which follows both linearity & time invariance property. | 5 min |
| 2 | **Division of the Topic**<br>-Convolution derivation of LTI sys<br>-IMPULSE INVARIENT PROPERTY OF LTI | 30min |
| 3. | **Conclusion**<br>In LTI sys linearity & time invariance are followed & o/p is obtained by convolution Of i/p signal with finite /infinite no of impulses. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is impulse function?<br>A1-a pulse of unity width going upward up to infinity.<br>Q2-Why only impulse function is taken for LTI sys?<br>A2-as it simply gives o/p of the system with out disturbing its actual i/ps. | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan10

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-II

| S. No. | Topic :-LINEAR TIME INVARIENT SYSTEM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>A Linear time invariant system is defined as the system which follows both linearity & time invariance property. | 5 min |
| 2 | **Division of the Topic**<br>-Convolution derivation of LTI sys<br>-IMPULSE INVARIENT PROPERTY OF LTI | 30min |
| 3. | **Conclusion**<br>In LTI sys linearity & time invariance are followed & o/p is obtained by convolution Of i/p signal with finite /infinite no of impulses. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is impulse function?<br>A1-a pulse of unity width going upward up to infinity.<br>Q2-Why only impulse function is taken for LTI sys?<br>A2-as it simply gives o/p of the system with out disturbing its actual i/ps. | 5 min |

Assignment to be given:-NIL

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan12

Semester:- VI  Class:-ECS  Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP  Unit:-II

| S. No. | Topic :-FIR & IIR SYSTEMS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>FINITE IMPULSE RESPONSE & INFINITE IMPULSE RESPONSE are ,if i/p are taken for finite time period & if i/p are taken for infinite period respectively. | 5 min |
| 2 | **Division of the Topic**<br>-Recursive & non recursive methods of realization<br>-Difference eqn for FIR & IIR SYS | 30min |
| 3. | **Conclusion**<br>In FIR sys, finite no of impulses are taken but for IIR infinite impulses are taken for system response. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is difference bw FIR & IIR SYSTEMS?<br>A1- if i/p are taken for finite time period & if i/p are taken for infinite period respectively.<br>Q2-How will you realize FIR & IIR sys?<br>A2-DIRECT 1, DIRECT2 & CASCADE FORMS | 5 min |

Assignment to be given:-DESCRIBE THE FIR & IIR FILTERS WITH PROPER EXAMPLE.

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan13

Semester:- VI Class:-ECS Course Code:-EC-614-F Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-II

| S. No. | Topic :-FIR & IIR SYSTEMS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>FINITE IMPULSE RESPONSE & INFINITE IMPULSE RESPONSE are ,if i/p are taken for finite time period & if i/p are taken for infinite period respectively. | 5 min |
| 2 | **Division of the Topic**<br>-Recursive & non recursive methods of realization<br>-Difference eqn for FIR & IIR SYS | 30min |
| 3. | **Conclusion**<br>In FIR sys, finite no of impulses are taken but for IIR infinite impulses are taken for system response. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is difference bw FIR & IIR SYSTEMS?<br>A1- if i/p are taken for finite time period & if i/p are taken for infinite period respectively.<br>Q2-How will you realize FIR & IIR sys?<br>A2-DIRECT 1, DIRECT2 & CASCADE FORMS | 5 min |

Assignment to be given:-DESCRIBE THE FIR & IIR FILTERS WITH PROPER EXAMPLE.

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY

# Lecture Plan-14

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-Z-TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>DEFINATION OF Z-TRANSFORM: is to transform a time domain signal into its complex plane X(z). | 5 min |
| 2 | **Division of the Topic**<br>-One sided Z-Transform ,two sided Z-Transform<br>-Causal & anti-causal Z-Transform | 30min |
| 3. | **Conclusion**<br>Z-Transform is used in analysis of discrete time signals & LTI systems | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by Z-Transform & why it is used?<br>A1-Z-TRANSFORM: is to transform a time domain signal into its complex plane X(z).<br>Q2-why Z-TRANSFORM is used?<br>A2- Z-Transform is used in analysis of discrete time signals & LTI systems | 5 min |

**Assignment to be given**:-NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-15

Semester:- VI  Class:-ECS  Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP  Unit:-IV

| S. No. | Topic :-PROPERTIES OF Z-TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>PROPERTIES OF Z- TRANSFORM & NUMERICAL BASED ON ITARE VERY USEFUL IN ANALYSING COMPLICATED Z-PLANE PROBLEMS. | 5 min |
| 2 | **Division of the Topic**<br>-Properties: LINEARITY<br>-TIME REVERSAL, TIME REVERSAL & SCALING,<br>-DIFFERENTIATION<br>I-NITIAL & FINAL VALUE THEORM | 30min |
| 3. | **Conclusion**<br>By using properties of Z-Transform we can have Z-Transform easily & directly. | 10 min |
| 4 | **Question / Answer**<br>Find the Z-Transform using property:<br>Q1-X[n]=u[-n]<br>A1-1/1-z<br>Q2- n^2 u(n)<br>A2-z^-1* (1+z^-1)/(1-z^-1) | 5 min |

**Assignment to be given**:-NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-16

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-PROPERTIES OF Z-TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>PROPERTIES OF Z- TRANSFORM & NUMERICAL BASED ON ITARE VERY USEFUL IN ANALYSING COMPLICATED Z-PLANE PROBLEMS. | 5 min |
| 2 | **Division of the Topic**<br>-Properties: LINEARITY<br>-TIME REVERSAL, TIME REVERSAL & SCALING,<br>-DIFFERENTIATION<br>I-NITIAL & FINAL VALUE THEORM | 30min |
| 3. | **Conclusion**<br>By using properties of Z-Transform we can have Z-Transform easily & directly. | 10 min |
| 4 | **Question / Answer**<br>Find the Z-Transform using property:<br>Q1-X[n]=u[-n]<br>A1-1/1-z<br>Q2- n^2 u(n)<br>A2-z^-1* (1+z^-1)/(1-z^-1) | 5 min |

**Assignment to be given**:-NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-17

Semester:- VI    Class:-ECS    Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP    Unit:-IV

| S. No. | Topic :-INVERSE Z-TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>FINDING INVERSE Z-TRANSFORM & NUMERICALS BASED ON IT TO GET THE ORIGINAL SEQUENCE AGAIN. | 5 min |
| 2 | **Division of the Topic**<br>-LONG DIVISION METHOD<br>-PARTIAL FRACTION EXPANSION METHOD | 30min |
| 3. | **Conclusion**<br>For finding the sequence again, we use long division & partial fraction methods which again gives us Seq X(n). | 10 min |
| 4 | **Question / Answer**<br>Q-Define the INVERSE Z-Transform :<br>A-For finding the sequence again, we use long division & partial fraction methods which again gives us Seq X(n). | 5 min |

**Assignment to be given**:- Find the INVERSE Z-Transform :
(a) X(z)= ln (1+a/z) ,z >a.
(b) X(z) = z/ (z-2) * (z-3)

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-18

Semester:- VI  Class:-ECS  Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP  Unit:-IV

| S. No. | Topic :-INVERSE Z-TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>FINDING INVERSE Z-TRANSFORM & NUMERICALS BASED ON IT TO GET THE ORIGINAL SEQUENCE AGAIN. | 5 min |
| 2 | **Division of the Topic**<br>-LONG DIVISION METHOD<br>-PARTIAL FRACTION EXPANSION METHOD | 30min |
| 3. | **Conclusion**<br>For finding the sequence again, we use long division & partial fraction methods which again gives us Seq X(n). | 10 min |
| 4 | **Question / Answer**<br>Q-Define the INVERSE Z-Transform :<br>A-For finding the sequence again, we use long division & partial fraction methods which again gives us Seq X(n). | 5 min |

**Assignment to be given**:- Find the INVERSE Z-Transform :
(a) $X(z)= \ln (1+a/z)$ ,z >a.
(b) $X(z) = z/ (z-2) * (z-3)$

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-19

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP Unit:-IV

| S. No. | Topic :-APPLICATIONS OF Z-TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Z –Transform plays an important role in analysis & representation of discrete time LTI system. | 5 min |
| 2 | **Causality of discrete time LTI sys**<br>-Stability of D.T.LTI SYS<br>-Linear constant coefficient difference eqn<br>-Determination of poles zeros of Z-transform | 30min |
| 3. | **Conclusion**<br>Properties of discrete time LTI sys can be directly related to their transfer function & their characteristics. Hence by Z-transform we can check for causality, stability & poles zeros. | 10 min |
| 4 | **Question / Answer**<br>Q1-What do you mean by transfer function?<br>A1-It is defined as the ratio of output of the sys to the input of sys.<br>Q2-What is the condition for stability of sys?<br>A2--sum of h(k)<infinite & ROC OF TRANSFER FN H(z) lie inside the unit circle. | 5 min |

**Assignment to be given**:- Solve the transfer function of discrete time LTI sys by linear constant diff eqn:
Y(n)= ½* y(n-1)+x(n)+1/3*x (n-1)

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-20

Semester:- VI    Class:-ECS    Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP    Unit:-IV

| S. No. | Topic :-APPLICATIONS OF Z-TRANSFORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Z –Transform plays an important role in analysis & representation of discrete time LTI system. | 5 min |
| 2 | **Causality of discrete time LTI sys**<br>-Stability of D.T.LTI SYS<br>-Linear constant coefficient difference eqn<br>-Determination of poles zeros of Z-transform | 30min |
| 3. | **Conclusion**<br>Properties of discrete time LTI sys can be directly related to their transfer function & their characteristics. Hence by Z-transform we can check for causality, stability & poles zeros. | 10 min |
| 4 | **Question / Answer**<br>Q1-What do you mean by transfer function?<br>A1-It is defined as the ratio of output of the sys to the input of sys.<br>Q2-What is the condition for stability of sys?<br>A2--sum of h(k)<infinite & ROC OF TRANSFER FN H(z) lie inside the unit circle. | 5 min |

**Assignment to be given**:- Solve the transfer function of discrete time LTI sys by linear constant diff eqn:
$Y(n)= ½* y(n-1)+x(n)+1/3*x (n-1)$

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-21

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP Unit:-III

| S. No. | Topic :-SAMPLING THEORM | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>A band limited signal x(t) having a finite energy which has no frequency components higher than Fh hertz can be completely described & reconstructed from its samples taken at rate of 2Fh samples per second. | 5 min |
| 2 | **Division of the Topic**<br>-Sampling theorem<br>-Nquist rate<br>-Frequency domain representation of sampled signal | 30min |
| 3. | **Conclusion**<br>Sampling is done to convert a continuous time signal into discrete time signal & minimum frequency for sampling is 2*f(max) of signal. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by nquist rate?<br>A1-minimum rate of sampling process.<br>Q2-Various fields where sampling is required?<br>A2-voice processing, image compression, speech signal analysis etc | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-22

Semester:- VI Class:-ECS Course Code:-EE-407-E
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP Unit:-III

| S. No. | Topic :-RECONSTRUCTION OF SAMPLED SIGNAL | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>.RECONSTRUCTION is getting original signal back from sampled signal &sufficient no of samples must be taken so that reconstruction results in whole signal. | 5 min |
| 2 | **Division of the Topic**<br>-CONDITIONS FOR SAMPLING.<br>-RECONSTRUCTION PRINCIPAL.<br>-LINEAR INTERPOLATION. | 30min |
| 3. | **Conclusion**<br>Hence frequency domain sampled signal can be converted into time domain by linear interpolation in which the specific pts of sampled signal are joined. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by reconstruction?<br>A1-getting original signal back from sampled signal.<br>Q2-What re the various conditions for sampling to be done?<br>A2- sufficient no of samples must be taken so that reconstruction resuts in whole signal. | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-23

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-III

| S. No. | Topic :-CHANGING THE SAMPLING RATE | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>DECIMATION is the method of reducing the sampling rate & INTERPOLATION is increasing the sampling rate. | 5 min |
| 2 | **Division of the Topic**<br>-DECIMATION<br>-INTERPOLATION<br>-Filter designing of interpolation & decimation | 30min |
| 3. | **Conclusion**<br>Sampling rate alteration is done by rational factor (I/D) with the help of interpolation & decimation as per need. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by sampling rate alteration?<br>A1-changing sampling rate according to our need.<br>Q2-What re the various conditions for decimation & interpolation?<br>A2-decimation filter is used to band limited signal before decimation operation. | 5 min |

Assignment to be given:-EXPLAIN THE PROCESS OF DECIMATION & INTERPOLATION.

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

# Lecture Plan24

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-III

| S. No. | Topic :-RECONSTRUCTION OF SAMPLED SIGNAL | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>.RECONSTRUCTION is getting original signal back from sampled signal &sufficient no of samples must be taken so that reconstruction results in whole signal. | 5 min |
| 2 | **Division of the Topic**<br>-CONDITIONS FOR SAMPLING.<br>-RECONSTRUCTION PRINCIPAL.<br>-LINEAR INTERPOLATION. | 30min |
| 3. | **Conclusion**<br>Hence frequency domain sampled signal can be converted into time domain by linear interpolation in which the specific pts of sampled signal are joined. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by reconstruction?<br>A1-getting original signal back from sampled signal.<br>Q2-What re the various conditions for sampling to be done?<br>A2- sufficient no of samples must be taken so that reconstruction resuts in whole signal. | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan -26

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-III

| S. No. | Topic :-DISCRETE TIME PROCESSING OF CONTINOUS TIME SIGNALS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Discrete time processing is done by transforming the continuous time signal into small intervals by sampling the signal into discrete intervals. | 5 min |
| 2 | **Division of the Topic**<br>-Time varying digital filter structures.<br>-Interfacing of digital systems with different sampling rate. | 30min |
| 3. | **Conclusion**<br>Sampling rate alteration is done by rational factor (I/D) | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by discrete time processing?<br>A1- Discrete time processing is done by transforming the continuous time signal into small intervals by sampling the signal into discrete intervals. | 5 min |

**Assignment to be given**:-GIVEN IN LECTURE NO. 11

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan -27

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-III

| S. No. | Topic :-DISCRETE TIME PROCESSING OF CONTINOUS TIME SIGNALS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Discrete time processing is done by transforming the continuous time signal into small intervals by sampling the signal into discrete intervals. | 5 min |
| 2 | **Division of the Topic**<br>-Time varying digital filter structures.<br>-Interfacing of digital systems with different sampling rate. | 30min |
| 3. | **Conclusion**<br>Sampling rate alteration is done by rational factor (I/D) | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by discrete time processing?<br>A1- Discrete time processing is done by transforming the continuous time signal into small intervals by sampling the signal into discrete intervals. | 5 min |

**Assignment to be given**:-GIVEN IN LECTURE NO. 11

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan28

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha
Subject:-DSP Unit:-III

| S. No. | Topic :-RECONSTRUCTION OF SAMPLED SIGNAL | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>.RECONSTRUCTION is getting original signal back from sampled signal &sufficient no of samples must be taken so that reconstruction results in whole signal. | 5 min |
| 2 | **Division of the Topic**<br>-CONDITIONS FOR SAMPLING.<br>-RECONSTRUCTION PRINCIPAL.<br>-LINEAR INTERPOLATION. | 30min |
| 3. | **Conclusion**<br>Hence frequency domain sampled signal can be converted into time domain by linear interpolation in which the specific pts of sampled signal are joined. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by reconstruction?<br>A1-getting original signal back from sampled signal.<br>Q2-What re the various conditions for sampling to be done?<br>A2- sufficient no of samples must be taken so that reconstruction resuts in whole signal. | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-30

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-INTRODUCTION TO FIR & IIR FILTERS. | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Digital filter is one in which both the i/p & o/p are discrete time signals. FIR & IIR Both are digital type of filter s based on FIR & IIR systems. | 5 min |
| 2 | **Division of the Topic**<br>-MAJOR CONSIDERATIONS IN USING DIGITAL FILTERS<br>-DIFFERENCE BW FIR & IIR FILTERS | 30min |
| 3. | **Conclusion**<br>IIR filters do not have linear phase & these are used for some phase distortion, these are stable & less flexible realized by recursive methods. FIR filters have linear phase , greater flexibility, more immunity to noise & realized by non recursive methods. | 10 min |
| 4 | **Question / Answer**<br>Q1-what is filtering?<br>A1-it is a process by which the frequency spectrum of signal can be modified or manipulated to get some result.<br>Q2-what is digital filter?<br>A2- Digital filter is one in which both the i/p & o/p are discrete time signals | 5 min |

**Assignment to be given:-** NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-31

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-INTRODUCTION TO FIR & IIR FILTERS. | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>Digital filter is one in which both the i/p & o/p are discrete time signals. FIR & IIR Both are digital type of filter s based on FIR & IIR systems. | 5 min |
| 2 | **Division of the Topic**<br>-MAJOR CONSIDERATIONS IN USING DIGITAL FILTERS<br>-DIFFERENCE BW FIR & IIR FILTERS | 30min |
| 3. | **Conclusion**<br>IIR filters do not have linear phase & these are used for some phase distortion, these are stable & less flexible realized by recursive methods. FIR filters have linear phase , greater flexibility, more immunity to noise & realized by non recursive methods. | 10 min |
| 4 | **Question / Answer**<br>Q1-what is filtering?<br>A1-it is a process by which the frequency spectrum of signal can be modified or manipulated to get some result.<br>Q2-what is digital filter?<br>A2- Digital filter is one in which both the i/p & o/p are discrete time signals | 5 min |

**Assignment to be given:-** NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-32

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-REALIZATION OF DIGITAL FILTERS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>For realization of digital filters the system function H(z) is specified & then its realization is possible. For implementing the specified difference equation of a filter addition , delay & multiplication by a constant. | 5 min |
| 2 | **Division of the Topic**<br>- BASIC BLOCK DIAGRAM REPRESENTATION<br>- DIRECT FORM 1 & DIRECT FORM 2<br>- TRANSPOSED FORM REALIZATION<br>-CASCADE FORM STRUCTURE | 30min |
| 3. | **Conclusion**<br>Due to finite precision arithmetic, sensitivity of the coefficients to quantization effects increases with the order of filter. this sensitivity may change the coefficient values & hence the frequency response causing the filter to be unstable. To overcome this cascade & parallel form realization is used. | 10 min |
| 4 | **Question / Answer**<br>Q1-what are advantages of block diagram representation?<br>A1-(a) computational algorithm can be easily drawn.<br>(b) hardware requirements can be easily developed.<br>(c) Relationship bw o/p & i/p can be easily determined<br>Q2-what is digital filter?<br>A2- Digital filter is one in which both the i/p & o/p are discrete time signals | 5 min |

**Assignment to be given:-** NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-33

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-REALIZATION OF DIGITAL FILTERS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>For realization of digital filters the system function H(z) is specified & then its realization is possible. For implementing the specified difference equation of a filter addition , delay & multiplication by a constant. | 5 min |
| 2 | **Division of the Topic**<br>- BASIC BLOCK DIAGRAM REPRESENTATION<br>- DIRECT FORM 1 & DIRECT FORM 2<br>- TRANSPOSED FORM REALIZATION<br>-CASCADE FORM STRUCTURE | 30min |
| 3. | **Conclusion**<br>Due to finite precision arithmetic, sensitivity of the coefficients to quantization effects increases with the order of filter. this sensitivity may change the coefficient values & hence the frequency response causing the filter to be unstable. To overcome this cascade & parallel form realization is used. | 10 min |
| 4 | **Question / Answer**<br>Q1-what are advantages of block diagram representation?<br>A1-(a) computational algorithm can be easily drawn.<br>(d) hardware requirements can be easily developed.<br>(e) Relationship bw o/p & i/p can be easily determined<br>Q2-what is digital filter?<br>A2- Digital filter is one in which both the i/p & o/p are discrete time signals | 5 min |

**Assignment to be given:-** NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-35

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-ANALYSIS OF FINITE WIRDLENGTH | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>In IIR filters the impulse response sequence is of infinite duration so impulse response sequence of it is of zero terms. The design methods of it are nonlinear. FIR filters are realized from analog filters. | 5 min |
| 2 | **Division of the Topic**<br>-IIR FILTERS<br>-FIR FILTERS<br>-FINITE WORDLENGTH EFFECT | 30min |
| 3. | **Conclusion**<br>IIR filters do not have linear phase & these are used for some phase distortion, these are stable & less flexible realized by recursive methods. IT involves transformation of analog domain into digital domain<br>H(n)= H(m-n-1) | 10 min |
| 4 | **Question / Answer**<br>Q1-where poles of IIR filter lies?<br>A1- on an ellipse<br>Q2-what is backward difference?<br>A2- to convert analog filter into digital filter. | 5 min |

**Assignment to be given**:- DESCRIBE THE VARIOUS DESIGNING METHODS OF IIR FILTERS.

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-36

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-ANALYSIS OF FINITE WIRDLENGTH | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>In IIR filters the impulse response sequence is of infinite duration so impulse response sequence of it is of zero terms. The design methods of it are nonlinear. FIR filters are realized from analog filters. | 5 min |
| 2 | **Division of the Topic**<br>-IIR FILTERS<br>-FIR FILTERS<br>-FINITE WORDLENGTH EFFECT | 30min |
| 3. | **Conclusion**<br>IIR filters do not have linear phase & these are used for some phase distortion, these are stable & less flexible realized by recursive methods. IT involves transformation of analog domain into digital domain<br>H(n)= H(m-n-1) | 10 min |
| 4 | **Question / Answer**<br>Q1-where poles of IIR filter lies?<br>A1- on an ellipse<br>Q2-what is backward difference?<br>A2- to convert analog filter into digital filter. | 5 min |

**Assignment to be given**:- DESCRIBE THE VARIOUS DESIGNING METHODS OF IIR FILTERS.

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-37

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-APPLICATIONS OF DSP | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>DSP tech are used in various areas as availability of high resolution spectral density, accuracy,. Dsp chips which occupy small space, low cost, less power consuming & highly reliable. | 5 min |
| 2 | **Division of the Topic**<br>-SPEECH SIGNAL ANALYSIS<br>-IMAGE PROCESSING<br>-VOICE PRIVACY<br>-RADIO RANGE RESOLUTION<br>-DOPPLER FILTERING<br>-IMAGE COMPRESSION | 30min |
| 3. | **Conclusion**<br>DSP is very popular in all fields availability of high resolution spectral density, accuracy,. Dsp chips which occupy small space, low cost, less power consuming & highly reliable. | 10 min |
| 4 | **Question / Answer**<br>Q1-what is channel vocoder?<br>A1-it is voice analysis synthesis system.<br>Q2-what is range of voice signal frequency?<br>A2- 300- 3000 hz | 5 min |

**Assignment to be given**:- NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-38

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-APPLICATIONS OF DSP | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>DSP tech are used in various areas as availability of high resolution spectral density, accuracy,. Dsp chips which occupy small space, low cost, less power consuming & highly reliable. | 5 min |
| 2 | **Division of the Topic**<br>-SPEECH SIGNAL ANALYSIS<br>-IMAGE PROCESSING<br>-VOICE PRIVACY<br>-RADIO RANGE RESOLUTION<br>-DOPPLER FILTERING<br>-IMAGE COMPRESSION | 30min |
| 3. | **Conclusion**<br>DSP is very popular in all fields availability of high resolution spectral density, accuracy,. Dsp chips which occupy small space, low cost, less power consuming & highly reliable. | 10 min |
| 4 | **Question / Answer**<br>Q1-what is channel vocoder?<br>A1-it is voice analysis synthesis system.<br>Q2-what is range of voice signal frequency?<br>A2- 300- 3000 hz | 5 min |

**Assignment to be given**:- NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-39

Semester:- VI | Class:-ECS | Course Code:-EC-614-F

Subject:-DSP | Unit:-IV

| S. No. | Topic :-MULTIRATE DSP | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>. MULTIRATE DSP is changing the sampling rate DECIMATION is the method of reducing the sampling rate & INTERPOLATION is increasing the sampling rate | 5 min |
| 2 | **Division of the Topic**<br>-DECIMATION<br>-INTERPOLATION | 30min |
| 3. | **Conclusion**<br>Sampling rate alteration is done by rational factor (I/D) with the help of interpolation & decimation as per need. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by sampling rate alteration?<br>A1-changing sampling rate according to our need.<br>Q2-What re the various conditions for decimation & interpolation?<br>A2-decimation filter is used to band limited signal before decimation operation | 5 min |

**Assignment to be given:-** NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

# Lecture Plan-40

Semester:- VI Class:-ECS Course Code:-EC-614-F

Subject:-DSP Unit:-IV

| S. No. | Topic :-SAMPLING RATE CONVERSION | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>A band limited signal x(t) having a finite energy which has no frequency components higher than Fh hertz can be completely described & reconstructed from its samples taken at rate of 2Fh samples per second. | 5 min |
| 2 | **Division of the Topic**<br>-Sampling theorem<br>-Nquist rate<br>-Frequency domain representation of sampled signal | 30min |
| 3. | **Conclusion**<br>Sampling is done to convert a continuous time signal into discrete time signal & minimum frequency for sampling is 2*f(max) of signal. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by nquist rate?<br>A1-minimum rate of sampling process.<br>Q2-Various fields where sampling is required?<br>A2-voice processing, image compression, speech signal analysis etc | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan-41

Semester:- VI | Class:-ECS | Course Code:-EC-614-F

Subject:-DSP | Unit:-IV

| S. No. | Topic :-SAMPLING RATE CONVERSION | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>A band limited signal x(t) having a finite energy which has no frequency components higher than Fh hertz can be completely described & reconstructed from its samples taken at rate of 2Fh samples per second. | 5 min |
| 2 | **Division of the Topic**<br>-Sampling theorem<br>-Nquist rate<br>-Frequency domain representation of sampled signal | 30min |
| 3. | **Conclusion**<br>Sampling is done to convert a continuous time signal into discrete time signal & minimum frequency for sampling is 2*f(max) of signal. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by nquist rate?<br>A1-minimum rate of sampling process.<br>Q2-Various fields where sampling is required?<br>A2-voice processing, image compression, speech signal analysis etc | 5 min |

Assignment to be given:-NIL

Reference Readings:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan42

Semester:- VI | Class:-ECS | Course Code:-EC-614-F

Subject:-DSP | Unit:-IV

| S. No. | Topic :-MULTIRATE DSP,INTERPOLATORS,DECIMATORS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>. MULTIRATE DSP is changing the sampling rate DECIMATION is the method of reducing the sampling rate & INTERPOLATION is increasing the sampling rate | 5 min |
| 2 | **Division of the Topic**<br>-DECIMATION<br>-INTERPOLATION | 30min |
| 3. | **Conclusion**<br>Sampling rate alteration is done by rational factor (I/D) with the help of interpolation & decimation as per need. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by sampling rate alteration?<br>A1-changing sampling rate according to our need.<br>Q2-What re the various conditions for decimation & interpolation?<br>A2-decimation filter is used to band limited signal before decimation operation | 5 min |

**Assignment to be given:-** NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN

Doc. No.: DCE/0/15
Revision :00

# Lecture Plan42

Semester:- VI Class:-ECS Course Code:-EC-614-F
Faculty Name: Akanksha Kulshreshtha

Subject:-DSP Unit:-IV

| S. No. | Topic :-MULTIRATE DSP,INTERPOLATORS,DECIMATORS | Time Allotted:- |
|---|---|---|
| 1. | **Introduction**<br>. MULTIRATE DSP is changing the sampling rate DECIMATION is the method of reducing the sampling rate & INTERPOLATION is increasing the sampling rate | 5 min |
| 2 | **Division of the Topic**<br>-DECIMATION<br>-INTERPOLATION | 30min |
| 3. | **Conclusion**<br>Sampling rate alteration is done by rational factor (I/D) with the help of interpolation & decimation as per need. | 10 min |
| 4 | **Question / Answer**<br>Q1-What is meant by sampling rate alteration?<br>A1-changing sampling rate according to our need.<br>Q2-What re the various conditions for decimation & interpolation?<br>A2-decimation filter is used to band limited signal before decimation operation | 5 min |

**Assignment to be given**:- NIL

**Reference Readings**:-VALLABHRAJ, OPPENHELM & WILLHSKY,FAROOQ HUSSIAN