# *DSP*

# *COURSE FILE*

# Contents

**Course file Contents:**

## 1. COVER PAGE

GEETHANJALI COLLEGE OF ENGINEERING AND TECHNOLOGY

DEPARTMENT OF Electronics and Communication Engineering

(Name of the Subject / Lab Course) : Digital Signal Processing

JNTU CODE -56027 Programme : *UG*

Branch: ECE Version No : *02*

Year: III *Year ECE ( C )* Document Number: GCET/ECE/DSP/02

Semester: II No. of pages :

Classification status (Unrestricted / Restricted )

Distribution List :

Prepared by : 1) Name : 1) Name : M.UMARANI

2) Sign : 2) Sign :

3) Design : Asst.Prof 3) Design :Asst.Prof

4) Date : 11/11/13 4) Date :18/11/15

| Verified by : | * For Q.C Only. |
|---|---|
| 1) Name : | 1) Name : |
| 2) Sign : | 2) Sign : |
| 3) Design : | 3) Design : |
| 4) Date : | 4) Date : |

Approved by : (HOD ) 1) Name : Dr.P.SRIHARI

2) Sign :

3) Date :



## 2. SYLLABUS

GEETHANJALI COLLEGE OF ENGINEERING & TECHNOLOGY

**Cheeryal, Keesara (M), R.R.Dist.**

**DEPARTMENT OF ECE**

SYLLABUS

# (A60421)Digital Signal Processing

**UNIT I: INTRODUCTION**

Introduction to Digital Signal Processing, Discrete time signals and sequences, Linear shift invariant Systems, Stability and Causality, Linear constant coefficient difference equations, Frequency domain representation of discrete time signals and systems.

**REALIZATION OF DIGITAL FILTERS:**

Applications of Z transforms, Solution of difference equations of Digital filters, System function, Stability Criterion, Frequency Response of stable systems, Realization of digital filters- Direct, Canonic, Cascade and Parallel forms.

**UNIT II: DISCRETE FOURIER SERIES**

DFS Representation of Periodic Sequences, Properties of discrete Fourier Series, Discrete Fourier Transforms, Properties of DFT, Linear Convolution of Sequences using DFT, Computation of DFT: Over-Lap Add Method, Over-Lap Save Method, Relation between DTFT, DFS, DFT and Z transform.

**FAST FOURIER TRANSFORMS**

Fast Fourier Transforms, Radix- 2 Decimation-in-Time and Decimation-in-Frequency FFT Algorithms, Inverse FFT, FFT with General Radix-N.

**UNIT III:**
**IIR DIGITAL FILTERS**

Analog filter approximations - Butter worth and Chebyshev, design IIR Digital Filters from Analog filters, Step and Impulse InvariantTechniques,Bilinear Transformation Method, Spectral transformations.

**UNIT IV: FIR DIGITAL FILTERS**

Characteristics of FIR digital filters, Frequency response, Design of FIR digital filters:Fourier Method, Digital Filters using Window Techniques, Frequency Sampling Technique, Comparison of IIR & FIR filters.

**UNIT V: MULTIRATE DIGITAL SIGNAL PROCESSING**

Introduction, Down Sampling, Decimation, Upsampling, Interpolation, Sampling Rate Conversion.

Finite word length effects: Limit cycles, Overflow oscillations, Round off Noise in IIR Digital Filters, Computational output Round off noise, Methods to prevent Overflow, Tradeoff between Round off and Over flow Noise, Dead Band Effects.

## 3. VISION OF THE DEPARTMENT

To impart quality technical education in Electronics and Communication Engineering emphasizing analysis, design/synthesis and evaluation of hardware/embedded software using various Electronic Design Automation (EDA) tools with accent on creativity, innovation and research thereby producing competent engineers who can meet global challenges with societal commitment.

## 4. MISSION OF THE DEPARTMENT

i. To impart quality education in fundamentals of basic sciences, mathematics, electronics and communication engineering through innovative teaching-learning processes.

ii. To facilitate Graduates define, design, and solve engineering problems in the field of Electronics and Communication Engineering using various Electronic Design Automation (EDA) tools.

iii. To encourage research culture among faculty and students thereby facilitating them to be creative and innovative through constant interaction with R & D organizations and Industry.

iv. To inculcate teamwork, imbibe leadership qualities, professional ethics and social responsibilities in students and faculty.

## 5. PEOS AND POS

**Program Educational Objectives of B. Tech (ECE) Program :**

I. To prepare students with excellent comprehension of basic sciences, mathematics and engineering subjects facilitating them to gain employment or pursue postgraduate studies with an appreciation for lifelong learning.

II. To train students with problem solving capabilities such as analysis and design with adequate practical skills wherein they demonstrate creativity and innovation that would enable them to develop state of the art equipment and technologies of multidisciplinary nature for societal development.

III. To inculcate positive attitude, professional ethics, effective communication and interpersonal skills which would facilitate them to succeed in the chosen profession exhibiting creativity and innovation through research and development both as team member and as well as leader.

**Program Outcomes of B.Tech ECE Program:**

1. An ability to apply knowledge of Mathematics, Science, and Engineering to solve complex engineering problems of Electronics and Communication Engineering systems.
2. An ability to model, simulate and design Electronics and Communication Engineering systems, conduct experiments, as well as analyze and interpret data and prepare a report with conclusions.
3. An ability to design an Electronics and Communication Engineering system, component, or process to meet desired needs within the realistic constraints such as economic, environmental, social, political, ethical, health and safety, manufacturability and sustainability.
4. An ability to function on multidisciplinary teams involving interpersonal skills.
5. An ability to identify, formulate and solve engineering problems of multidisciplinary nature.
6. An understanding of professional and ethical responsibilities involved in the practice of Electronics and Communication Engineering profession.
7. An ability to communicate effectively with a range of audience on complex engineering problems of multidisciplinary nature both in oral and written form.
8. The broad education necessary to understand the impact of engineering solutions in a global, economic, environmental and societal context.
9. Recognition of the need for, and an ability to engage in life-long learning and acquire the capability for the same.

10. A knowledge of contemporary issues involved in the practice of Electronics and Communication Engineering profession
11. An ability to use the techniques, skills and modern engineering tools necessary for engineering practice.
12. An ability to use modern Electronic Design Automation (EDA) tools, software and electronic equipment to analyze, synthesize and evaluate Electronics and Communication Engineering systems for multidisciplinary tasks.
13. Apply engineering and project management principles to one's own work and also to manage projects of multidisciplinary nature.

## 6. Course objectives and outcomes

**Course objectives:**

- This course will introduce the basic concepts and techniques for processing signals on a computer. By the end of the course, students will be familiar with the most important methods in DSP, including digital filter design, transform-domain processing and importance of Signal Processors.
- The course emphasizes intuitive understanding and practical implementations of the theoretical concepts.
- To produce graduates who understand how to analyze and manipulate digital signals and have the fundamental Mat lab programming knowledge to do so.

**Course Outcomes:**

**CO 1:** Able to obtain different Continuous and Discrete time signals.

**CO 2:** Able to calculate Z-transforms for discrete time signals and system functions.

**CO 3:** Ability to calculate discrete time domain and frequency domain of signals using discrete Fourier series and Fourier transform.

**CO 4:**.Ability to develop Fast Fourier Transform (FFT) algorithms for faster realization of signals and systems.

**CO 5:** Able to design Digital IIR filters from Analog filters using various techniques (Butterworth and

Chebyshev).

**CO 6:** Able to design Digital FIR filters using window techniques,Fouriour methods and frequency sampling technique..

**CO 7:** Ability to design different kinds of interpolator and decimator.

**CO 8:** Ability to demonstrate the impacts of finite word length effects in filter design.

## 7. BRIEF NOTE ON THE IMPORTANTANCE OF THE COURSE AND HOW IT FITS IN TO THE CURRICULAM

Digital Signal Processing (DSP) is concerned with the representation, transformation and manipulation of signals on a computer. After half a century advances, DSP has become an important field, and has penetrated a wide range of application systems, such as consumer electronics, digital communications, medical imaging and so on. With the dramatic increase of the processing capability of signal processing microprocessors, it is the expectation that the importance and role of DSP is to accelerate and expand.

Discrete-Time Signal Processing is a general term including DSP as a special case. This course will introduce the basic concepts and techniques for processing discrete-time signal. By the end of this course, the students should be able to understand the most important principles in DSP. The course emphasizes understanding and implementations of theoretical concepts, methods and algorithms.

## 8. PREREQUISITES, IF ANY

- Laplace Transforms
- Fourier Transforms
- Signals and systems

## 9. INSTRUCTIONAL LEARNING OUTCOMES

**UNIT-I (INTRODUCTION)**

1) Students can understand the concept of discrete time signals & sequences.
2) Analyze and implement digital signal processing systems in time domain.
3) They can solve linear constant coefficient difference equations.

4) They can understand Frequency domain representation of discrete time signals and systems.
5) They can understand the practical purpose of stability and causality.
6) To determine stability, causality for a given impulse response.
7) Understand how analog signals are represented by their discrete-time samples, and in what ways digital filtering is equivalent to analog filtering.
8) The basics of Z-transforms and its applications are studied.
9) Digital filters are realized using difference equations.
10) Calculate the response of applying a given input signal to a system described by a linear constant coefficient differential equation.

**UNIT-II (DISCRETE FOURIER SERIES)**

1) Ability to understand discrete time domain and frequency domain representation of signals and systems.
2) Compute convolution and the discrete Fourier transform (DFT) of discrete-time signals.
3) Analyze and implement digital systems using the DFT.
4) Ability to understand Discrete Fourier Series and Transforms and comparison with other transforms like Z transforms.
5) Ability to represent discrete-time signals in the frequency domain.
6) Calculate exponential Fourier series coefficients using properties of Fourier series
7) Graphically portray the magnitude and phase of the Fourier series coefficients versus ω.

**Fast Fourier Transforms**

8) Ability to develop Fast Fourier Transform algorithms for faster realization of signals and systems.
9) Ability to understand discrete time domain and frequency domain representation of signals and systems.
10) Analyze and implement digital systems using the FFT.
11) Describe how and why Fourier Transforms and Fourier series are related.

**UNIT-III (IIR DIGITAL FILTERS)**

1) The student is able to solve basic digital signal processing algorithms.
2) Assess signal acquisition, processing, and reconstruction.
3) Ability to understand the concepts of Digital filters IIR like Chebychev, Butterworth filters.
4) The student is able to solve the impulse and frequency response of IIR filters given as difference equations, transfer functions, or realization diagrams, and can present analyses of the aliasing and imaging effects based on the responses of the filters.
5) Learn the basic forms of IIR filters, and how to design filters with desired frequency responses.

**UNIT-IV (FIR DIGITAL FILTERS)**

1) Ability to understand the characteristics of linear-phase finite impulse response (FIR) filters
2) Ability to understand Digital Filters with special emphasis on realization of FIR and IIR filters.
3) Ability to design linear-phase FIR filters according to predefined specifications using the window and frequency sampling methods
4) Ability to understand the concepts of Digital FIR filters.

5) The student is able to specify and design respective frequency selective FIR and IIR filters using the most common methods.
6) The student is able to solve for the impulse and frequency responses of FIR and IIR filters given as difference equations, transfer functions, or realization diagrams, and can present analyses of the aliasing and imaging effects based on the responses of the filters.
7) Measure the effectiveness of FIR filters.

**UNIT-V (MULTIRATE DIGITAL SIGNAL PROCESSING)**

1) Ability to understand the concepts of sampling rate conversions, Decimation and Interpolation as part of Signal Processing techniques.
2) Able to explain how the multirate implementation of ADC and DAC converters works.
3) Able to describe basic sampling rate conversion algorithms.
4) Able to draw and describe different kinds of interpolator and decimator.
5) Able to analyze how the interpolated FIR filter works.
6) Able to do sampling rate conversion.

**Finite word length effects in Digital filters**

1) Ability to analyze the concepts of Signal scaling.
2) Ability to analyze the concepts Quantization errors in FFT algorithms.
3) Ability to analyze Number representation, Floating point numbers, quantization Noise, Overflow limit cycle oscillations, Signal scaling, Finite Word Length effects in FIR, Quantization errors in FFT algorithms.
4) The student is able to explain the impact of finite word length in filter design.
5) The student is able to design simple digital filters.
6) Concept of quantization Noise is analyzed and errors are identified.

## 10. <u>Course mapping with Programme Outcomes</u>:

**<u>Mapping of Course outcomes with Programme outcomes:</u>**

*When the course outcome weightage is < 40%, it will be given as moderately correlated (1).
*When the course outcome weightage is >40%, it will be given as strongly correlated (2).

| POs | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | **Digital signal Processing** |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| Digital Signal Processing | | **2** | **2** | | **2** | | | **2** | | | **2** | **2** | | |
| **CO 1:** Able to | | **2** | | | **2** | | | | | | | | | |

| | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| demonstrate different Analog and Discrete time signals. | | | | | | | | | | | | | | |
| **CO 2:** Ability to calculate discrete time domain and frequency domain of signals using discrete Fourier Series and Fourier transform. | | **2** | | | **2** | | | | | | **2** | **2** | | |
| **CO 3:** Ability to develop Fast Fourier Transform (FFT) algorithms for faster realization of signals and systems. | | **2** | **2** | | **2** | | | | | | **2** | **2** | | |
| **CO 4:** Able to calculate Z-transforms for discrete time signals and system functions. | | **2** | | | **2** | | | | | | **2** | **2** | | |
| **CO 5:** Able to design Digital IIR filters from Analog filters using various techniques (Butterworth and Chebyshev). | | **2** | **1** | | **2** | | | | | | **2** | **2** | **2** | |
| **CO 6:** Able to design Digital FIR filters using window techniques,Fouriour | | **2** | **1** | | **2** | | | | | | **2** | **2** | **2** | |

| | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| methods and frequency sampling technique. | | | | | | | | | | | | | | |
| **CO 7:** Ability to design different kinds of interpolator and decimator. | | **2** | | | **2** | | | | | | **2** | **2** | **2** | |
| **CO 8:** Ability to demonstrate the impacts of finite word length effects in filter design. | | **2** | | | **2** | | | | | | **2** | **2** | **2** | |



## 11. <u>Class Time Table</u>

To be attached

## 12. <u>Individual time Table</u>

**<u>Name:</u> Ms.M.Umarani ver: 1 w.e.f.: 07/12/2015 Load:18**

| Day | 1 | 2 | 3 | 4 | | 5 | 6 | 7 |
|---|---|---|---|---|---|---|---|---|
| Mon | DSP | | | | LUNCH | DSP lab | | |
| TUE | | DSP lab | | | | | DSP | |
| WED | | | | | | | | DSP |

| | | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| THUR | | | | | | | | |
| FRI | | | | | | DSP | | |
| SAT | | DSP(T) | | | | | Technical seminars | |



## 13. LECTURE SCHEDULE WITH METHODOLOGY BEING USED/ ADOPTED

| Sl. No. | Unit No. | Total number of periods | Date | Topic to be covered in One lecture | Regular/ Additional/ Missing | Teaching Aids used LCD /OHP/BB | Remarks |
|---|---|---|---|---|---|---|---|
| 1 | UNIT I | 16 | 07/12/15 | Introduction to Digital Signal processing | Regular | BB | |
| 2 | | | 30/12/14 | Discrete time signals and sequences | Regular | BB | |
| 3 | | | 31/12/14 | Linear shift invariant systems, stability and causality | Regular | BB | |
| 4 | | | 01/01/15 | Linear constant coefficient difference equations | Regular | BB | |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 5 | | | 02/01/15 | Frequency domain representation of discrete time signals and systems | Regular | BB/OHP | |
| | | | | **Wavelet Transforms** | **Missing** | BB/OHP | |
| 6 | | | 03/01/15 | Tutorial class-1 | | BB | |
| 7 | | | | Review of Z transforms | | | |
| 8 | | | | Applications of Z transforms ,solution of difference equations | | | |
| 9 | | | | Block diagram representation of linear constant coefficient | | | |
| 10 | | | | Basic structures of IIR systems, Transposed forms | | | |
| 11 | | | | Basic structure of FIR systems, System function | | | |
| 12 | | | | Tutorial class-2 | | | |
| 13 | | | | Structures of cascade and parallel forms | | | |
| 14 | | | | Solving university question paper / Revision | | | |
| 15 | | | | Assignment test-2 | | | |
| 16 | | | | **MID TEST 1** | | | |
| 17 | | | 05/01/15 | Properties of discrete Fourier series , DFS representation of periodic sequences | Regular | OHP | |
| 18 | | | 06/01/15 | Discrete Fourier transformers, Properties of DFT, | Regular | BB | |
| 19 | | | 07/01/15 | Linear convolution of sequences using DFT, | Regular | BB | |
| 20 | | | 08/01/15 | Computation of DFT | Regular | BB | |
| 21 | **UNIT II** | **14** | 10/01/15 | Relation between Z transform and DFS. | Regular | BB | |
| 22 | | | | **Applications of MAC** | **Additional** | BB/OHP/ LCD | |
| 23 | | | 10/01/15 | Tutorial class-3 | | BB/OHP/L CD | |
| 24 | | | 12/01/15 | Solving university question papers / Revision | | BB/OHP/L CD | |
| 25 | | | 13/01/15 | Assignment test-1 | | BB/OHP/L CD | |
| 26 | | | | Fast Fourier Transform | | | |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 27 | | | | Radix to decimation in time | | | |
| 28 | | | | decimation in frequency FFT algorithms | | | |
| | | | | Inverse FFT | | | |
| 29 | | | | FFT for composite N | | | |
| 30 | | | | Tutorial class-4 | | | |
| 31 | **UNIT III** | | 12/01/15 | Analog filter approximations - Butter worth | Regular | BB | |
| 32 | | | 15/01/15 | Analog filter approximations - chebyshev | Regular | OHP | |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 33 | | 10 | 15/01/15 | Design of IIR DIGITAL FILTERS from analog filters | Regular | OHP | |
| 34 | | | 19/01/15 | Impulse Invariance technique | Regular | OHP | |
| 35 | | | 20/01/15 | Bi-linear Transformation | Regular | OHP | |
| 36 | | | 20/01/15 | Design examples: Analog digital transformations | | BB/OHP | |
| | | | | **Speech processing** | | | |
| 37 | | | | Spectral Transformations | | | |
| 38 | | | | Solving university question paper | | | |
| 39 | | | | Revision | | | |
| 40 | | | | Tutorial class-5 | | | |
| 41 | **UNIT I V** | **10** | 30/01/15 | Concept of FIR filters | Regular | BB/OHP | |
| 42 | | | 31/01/15 | Frequency Response of FIR filters | Regular | BB/OHP | |
| 43 | | | 02/02/15 | Design of FIR digital filters using Fourier Method | Regular | BB/OHP | |
| 44 | | | 03/02/15 | Design of FIR digital filters using windows Techniques | Regular | BB/OHP | |
| 45 | | | 03/02/15 | Frequency sampling technique | **Additional** | BB/OHP | |
| 46 | | | 05/02/15 | Comparison of IIR & FIR filters | | BB | |
| 47 | | | 06/02/15 | Tutorial class-6 | Regular | BB/OHP | |
| 48 | | | 09/02/15 | Solving university question papers | Regular | BB/OHP | |
| 49 | | | 09/02/15 | Revision | Regular | BB/OHP | |
| 50 | | | | Assignment test-4 | | | |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 51 | | | 10/02/15 | Concept of Multirate signal Processing | Regular | BB/OHP | |
| 52 | | | 11/02/15 | Decimation | Regular | BB/OHP | |
| 53 | | | 12/02/15 | interpolation | | BB | |
| 54 | | | 15/02/15 | Sampling rate conversion | | BB | |
| 55 | | | 18/02/15 | Implementation of sampling rate conversion | | BB/OHP | |
| 56 | | | 19/02/15 | Multi stage implementation of sampling rate conversion | Regular | BB/OHP | |
| 57 | | | 20/02/15 | **DSP Processors** | Regular | BB/OHP | |
| 58 | | | 23/02/15 | Tutorial class-7 | Regular | BB/OHP | |
| 59 | | | 25/02/15 | Introduction to programmable DSPs, Multiplier and | Regular | BB/OHP | |
| 60 | | | 27/02/15 | Multiport memory, VLSI Architecture, Pipelining, | Regular | BB/OHP | |
| 61 | | | 28/02/15 | Special addressing modes, On – chip peripherals | **Missing** | BB/OHP | |
| 62 | | | 02/03/15 | Architecture of TMS320C5X- Introduction, bus structure | | BB | |
| 63 | | | 03/03/15 | Auxillary register, index Register, Auxillary Register Compare Register, Block move address register, memory mapped registers | Regular | BB/OHP | |
| 64 | | | 04/03/15 | program controller, some flags in status registers, On chip | Regular | BB/OHP | |
| | | | 06/03/15 | peripherals. | Regular | BB/OHP | |
| | | | 07/03/15 | Tutorial class-8 | Regular | BB/OHP | |
| | | | 09/02/15 | Solving university question papers / Revision | Regular | BB/OHP | |
| | | | 10/03/15 | | Regular | BB/OHP | |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| | | | 12/03/15 | | | | |
| | | | 13/03/15 | | | BB | |
| | | | 16/03/15 | | | BB | |
| | | | 18/03/15 | Assignment test-4 | | | |
| | | | 23/03/15 | Assignment test | | | |
| | | | 26/03/15 | Assignment test | | | |
| | | | | **MID TEST II** | | | |

**GUIDELINES:**

Distribution of periods:

- No. of classes required to cover JNTU syllabus : 46
- No. of classes required to cover Additional topics : 02
- No. of classes required to cover Assignment tests (for every 2 units 1 test) : 04
- No. of classes required to cover tutorials : 08
- No of classes required to solve University : 04
- Question papers : 04

Total periods : 64 .

### Introduction to the subject

In the class, starting from the basic definitions of a discrete-time signal, we will work our way through Fourier analysis, filter design, sampling, interpolation and quantization to build a DSP toolset complete enough to analyze a practical communication system in detail. Hands-on examples and demonstration will be routinely used to close the gap between theory and practice.

The digital signal processor can be programmed to perform a variety of signal processing operations, such as filtering, spectrum estimation, and other DSP algorithms. Depending on the speed and computational requirements of the application, the digital signal processor may be realized by a general purpose computer, minicomputer, special purpose DSP chip, or other digital hardware dedicated to performing a particular signal processing task.

## 14. DETAILED NOTES:

IIR filters are digital filters with infinite impulse response. Unlike FIR filters, they have the feedback (a recursive part of a filter) and are known as recursive digital filters therefore.

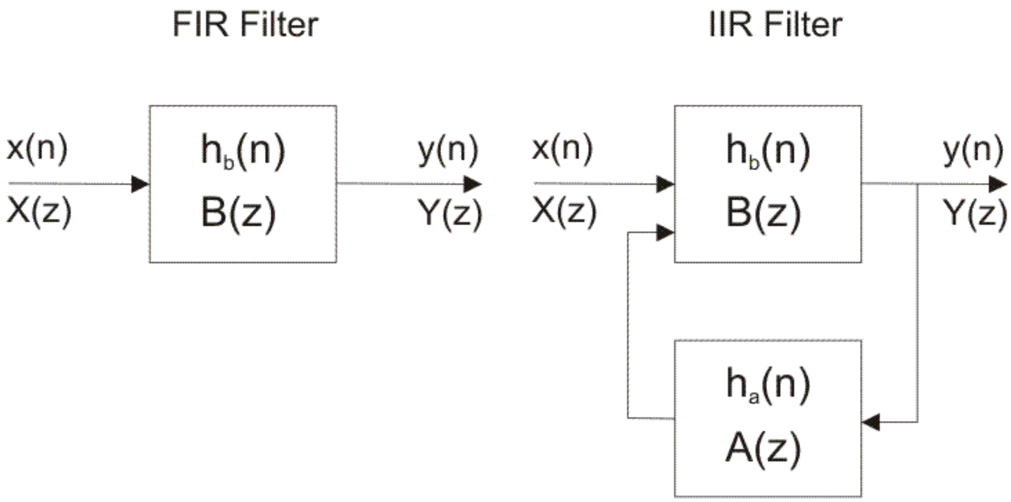


**Figure 3-1-1. Block diagrams of FIR and IIR filters**

For this reason IIR filters have much better frequency response than FIR filters of the same order. Unlike FIR filters, their phase characteristic is not linear which can cause a problem to the systems which need phase linearity. For this reason, it is not preferable to use IIR filters in digital signal processing when the phase is of the essence.

Otherwise, when the linear phase characteristic is not important, the use of IIR filters is an excellent solution.

There is one problem known as a potential instability that is typical of IIR filters only. FIR filters do not have such a problem as they do not have the feedback. For this reason, it is always necessary to check after the design process whether the resulting IIR filter is stable or not.

IIR filters can be designed using different methods. One of the most commonly used is via the reference analog prototype filter. This method is the best for designing all standard types of filters such as low-pass, high-pass, band-pass and band-stop filters.

This book describes the design method using reference analog prototype filter. Figure 3-1-2 illustrates the block diagram of this method.

**UNIT V - Multirate Digital Signal Processing**

The process of converting a signal from a given rate to a different rate is called **sampling rate conversion.** In turn,
systems that employ multiple sampling rates in the processing of digital signals are called **multirat~digital signal processing systems.**
Sampling rate conversion of a digital signal can be accomplished in one of two general methods. One method is to pass the digital signal through a DIA converter, filter it if necessary, and then to resample the resulting analog signal at the desired rate (i.e., to pass the analog signal through an AID converter). The second method is to perform the sampling rate conversion entirely in the digital domain.One apparent advantage of the first method is that the new sampling rate can be arbitrarily selected and need not have any special relationship to the old sampling rate. A major disadvantage, however, is the signal distortion, introduced by the **DIA** converter in the signal reconstruction, and by the quantization effects in the **AD** conversion. Sampling rate conversion performed in the digital domain avoids this major disadvantage.Here we describe sampling rate conversion and multirate signal processing in the digital domain. First we describe sampling rate conversion by a rational factor and present several methods for implementing the rate converter, including single-stage and multistage implementations. Then, we describe a method for sampling rate conversion by an arbitrary factor and discuss its implementation.Finally, we present several applications of sampling rate conversion in multirate signal processing systems, which include the implementation of narrowband filters,digital filter banks, and quadrature mirror filters. We also discuss the use of quadrature mirror filters in subband coding. transmultiplexers. and finally oversampling A/D and **D/A** converters.

**INTRODUCTION**

The process of sampling rate conversion in the digital domain can be viewed asa linear filtering operation, as illustrated . The input signal **x ( n )**is characterized by the sampling rate **F,** = 1/T, and the output signal **y(m)** is characterized by the sampling rate **F!** = l/T,., where **T,** and 7j, are the correspondingsampling intervals. In the main part of our treatment, the ratio **F, /Fl**isconstrained to be rational,where **D** and I are relatively prime integers. We shall show that the linear filteris characterized by a time-variant impulse response. denoted as **h(tl. m).** Hencethe input **x(?r)** and the output y(m) are related by the convolution summation fortime-variant systems.The sampling rate conversion process can also be understood from the pointof view of digital resampling of the same analog signal. Let **x ( r )** be the analogsignal that is sampled at the first rate **F,** to generate **x(t1).** The goal ofrate conversion is to obtain another sequence **j ~ ( n t )** directly from **x ( n ) .** whichis equal to the sampled values of **x t t )** at a second rate **F!.** As is depicted **.!*(m)** is a time-shifted version of **x ( n ) .** Such a time shift can be

**UNIT V - Multirate Digital Signal Processing**

The process of converting a signal from a given rate to a different rate is called **sampling rate conversion.** In turn,
systems that employ multiple sampling rates in the processing of digital signals are called **multirat~digital signal processing systems.**
Sampling rate conversion of a digital signal can be accomplished in one of two general methods. One method is to pass the digital signal through a DIA converter, filter it if necessary, and then to resample the resulting analog signal at the desired rate (i.e., to pass the analog signal through an AID converter). The second method is to perform the sampling rate conversion entirely in the digital domain.One apparent advantage of the first method is that the new sampling rate can be arbitrarily selected and need not have any special relationship to the old sampling rate. A major disadvantage, however, is the signal distortion, introduced by the **DIA** converter in the signal reconstruction, and by the quantization effects in the **AD** conversion. Sampling rate conversion performed in the digital domain avoids this major disadvantage.Here we describe sampling rate conversion and multirate signal processing in the digital domain. First we describe sampling rate conversion by a rational factor and present several methods for implementing the rate converter, including single-stage and multistage implementations. Then, we describe a method for sampling rate conversion by an arbitrary factor and discuss its implementation.Finally, we present several applications of sampling rate conversion in multirate signal processing systems, which include the implementation of narrowband filters,digital filter banks, and quadrature mirror filters. We also discuss the use of quadrature mirror filters in subband coding. transmultiplexers. and finally oversampling A/D and **D/A** converters.

**INTRODUCTION**

The process of sampling rate conversion in the digital domain can be viewed asa linear filtering operation, as illustrated . The input signal **x ( n )**is characterized by the sampling rate **F,** = 1/T, and the output signal **y(m)** is characterized by the sampling rate **F!** = l/T,., where **T,** and 7j, are the correspondingsampling intervals. In the main part of our treatment, the ratio **F, /Fl**isconstrained to be rational,where **D** and I are relatively prime integers. We shall show that the linear filteris characterized by a time-variant impulse response. denoted as **h(tl. m).** Hencethe input **x(?r)** and the output y(m) are related by the convolution summation fortime-variant systems.The sampling rate conversion process can also be understood from the pointof view of digital resampling of the same analog signal. Let **x ( r )** be the analogsignal that is sampled at the first rate **F,** to generate **x(t1).** The goal ofrate conversion is to obtain another sequence **j ~ ( n t )** directly from **x ( n ) .** whichis equal to the sampled values of **x t t )** at a second rate **F!.** As is depicted **.!*(m)** is a time-shifted version of **x ( n ) .** Such a time shift can be

## 4.1. Introduction

An FIR digital filter of order M may be implemented by programming the signal-flow-graph shown below. Its difference equation is:

$$y[n] = a_0 x[n] + a_1 x[n-1] + a_2 x[n-2] + \ldots + a_M x[n-M]$$

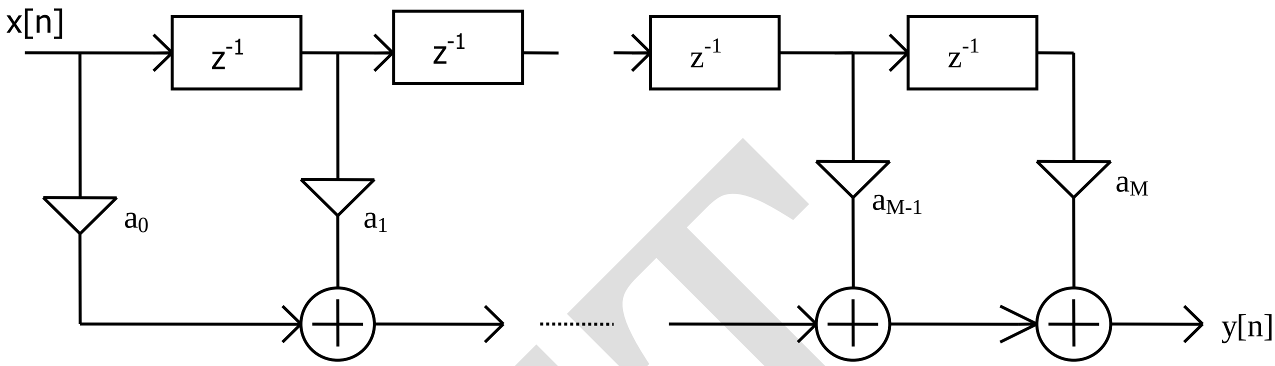


Fig. 4.1

Its impulse-response is {..., 0, ..., $\underline{a_0}$, $a_1$, $a_2$,..., $a_M$, 0, ...} and its frequency-response is the DTFT of the impulse-response, i.e.

$$\boxed{H(e^{j\Omega}) \quad = \quad \sum_{n=-\infty}^{\infty} h[n]e^{-j\Omega n} \quad = \quad \sum_{n=0}^{M} a_n e^{-j\Omega n}}$$

Now consider the problem of choosing the multiplier coefficients. $a_0$, $a_1$,..., $a_M$ such that $H(e^{j\Omega})$ is close to some desired or target frequency-response $H'(e^{j\Omega})$ say. The inverse DTFT of $H'(e^{j\Omega})$ gives the required impulse-response :

$$h'[n] = \frac{1}{2\pi}\int_{-\pi}^{\pi} H'(e^{j\Omega})e^{j\Omega n}d\Omega$$

The methodology is to use the inverse DTFT to get an impulse-response {h′[n]} & then realise some approximation to it Note that the DTFT formula is an integral, it has complex numbers and the range of integration is from -π to π, so it involves negative frequencies.

<u>Reminders about integration</u>

(1) If $x(t) = e^{at}$ then $\dfrac{dx}{dt} = ae^{at}$

$$\therefore \quad \int_{-\pi}^{\pi} x(t)dt \quad = \quad \int_{-\pi}^{\pi} e^{at}dt \quad = \quad \left[\frac{1}{a}e^{at}\right]_{-\pi}^{\pi} \quad = \quad \frac{1}{a}\left[e^{a\pi} - e^{-a\pi}\right]$$

**FINITE WORD LENGTH EFFECTS**

Practical digital filters must be implemented with finite precision numbers and arithmetic. As a result, both the filter coefficients and the filter input and output signals are in discrete form. This leads to four types of finite wordlength effects. Discretization (quantization) of the filter coefficients has the effect of perturbing the location of the filter poles and zeroes. As a result, the actual filter response differs slightly from the ideal response. This deterministic frequency response error is referred to as coefficient quantization error. The use of finite precision arithmetic makes it necessary to quantize filter calculations by rounding or truncation. Roundoffnoise is that errorin the filter output that resultsfrom rounding or truncating calculations within the filter. As the name implies, this error looks like low-level noise at the filter output. Quantization of the filter calculations also renders the filter slightly nonlinear. For large signals this nonlinearity is negligible and roundoff noise is the major concern. However, for recursive filters with a zero or constant input, this nonlinearity can cause spurious oscillations called limit cycles. With fixed-point arithmetic it is possible for filter calculations to overflow. The term overflow oscillation, sometimes also called adder overflow limit cycle, refers to a high-level oscillation that can exist in an otherwise stable filter due to the nonlinearity associated with the overflow of internal filter calculations. In this chapter, we examine each of these finite length effects. Both fixed-point and floatingpoint number representations are consider

**Limit Cycles**

A limit cycle, sometimes referred to as a multiplier roundoff limit cycle, is a low-level oscillation that can exist in an otherwise stable filter as a result of the nonlinearity associated with rounding (or truncating) internal filter calculations `. Limit cycles require recursion to exist and do not occur in nonrecursive FIR filters. c 1999 by CRC Press LLC As an example of a limit cycle, consider the second-order filter realized by y(n) = Qr7 8 y(n − 1) − 5 8 y(n − 2) + x(n) where Qr{ }represents quantization by rounding. This is stable filter with poles at 0.4375±j0.6585. Consider the implementation of this filter with 4-b (3-b and a sign bit) two's complement fixed-point arithmetic, zero initial conditions (y(−1) = y(−2) = 0), and an input sequence x(n) = 3 8 δ(n), where δ(n) is the unit impulse or unit sample. The following sequence is obtained; y(0) = Qr3 8 = 3 8 y(1) = Qr21 64 = 3 8 y(2) = Qr3 32 = 1 8 y(3) = Qr −1 8 = −1 8 y(4) = Qr− 3 16 = −1 8 y(5) = Qr− 1 32 = 0 y(6) = Qr5 64 = 1 8 (3.70) y(7) = Qr7 64 = 1 8 y(8) = Qr1 32 = 0 y(9) = Qr− 5 64 = −1 8 y(10) = Qr− 7 64 = −1 8 y(11) = Qr− 1 32 = 0 y(12) = Qr5 64 = 1 8 . . . Notice that while the input is zero except for the first sample, the output oscillates with amplitude 1/8 and period 6. Limit cycles are primarily of concern in fixed-point recursive filters. As long as floating-point filters are realized as the parallel or cascade connection of first- and second-order subfilters, limit cycles will generally not be a problem since limit cycles are practically not observable in first- and second-order systems implemented with 32-b floating-point arithmetic . It has been shown that such systems must have an extremely small margin of

## 15. ADDITIONAL TOPICS

### Additional/missing topics

- Speech processing
- Radar Signal Processing
- DSP Processors
- Pulse Code Modulation
- Correlation
- Geortzel algorithm
- FIR Least square design methods
- Multi stage implementation of sampling rate conversion

### 1. Correlation

The concept of correlation can best be presented with an example. Figure 7-13 shows the key elements of a radar system. A specially designed antenna transmits a short burst of radio wave energy in a selected direction. If the propagating wave strikes an object, such as the helicopter in this illustration, a small fraction of the energy is reflected back toward a radio receiver located near the transmitter. The transmitted pulse is a specific shape that we have selected, such as the triangle shown in this example. The received signal will consist of two parts: (1) a shifted and scaled version of the transmitted pulse, and (2) random noise, resulting from interfering radio waves, thermal noise in the electronics, etc. Since radio signals travel at a known rate, the speed of light, the shift between the transmitted and received pulse is a direct measure of the distance to the object being detected. This is the problem: given a signal of some known shape, what is the best way to determine where (or if) the signal occurs in *another* signal. Correlation is the answer.

Correlation is a mathematical operation that is very similar to convolution. Just as with convolution, correlation uses two signals to produce a third signal. This third signal is called the **cross-correlation** of the two input signals. If a signal is correlated with *itself*, the resulting signal is instead called the **autocorrelation**. The convolution machine was presented in the last chapter to show how convolution is performed. Figure 7-14 is a similar illustration of a **correlation machine**. The received signal, $x[n]$, and the cross-correlation signal, $y[n]$, are fixed on the page. The waveform we are looking for, $t[n]$, commonly called the **target** signal, is contained *within* the correlation machine. Each sample in $y[n]$ is calculated by moving the correlation machine left or right until it points to the sample being worked on. Next, the indicated samples from the received signal fall into the correlation machine, and are multiplied by the corresponding points in the target signal. The sum of these products then moves into the proper sample in the cross-correlation signal.

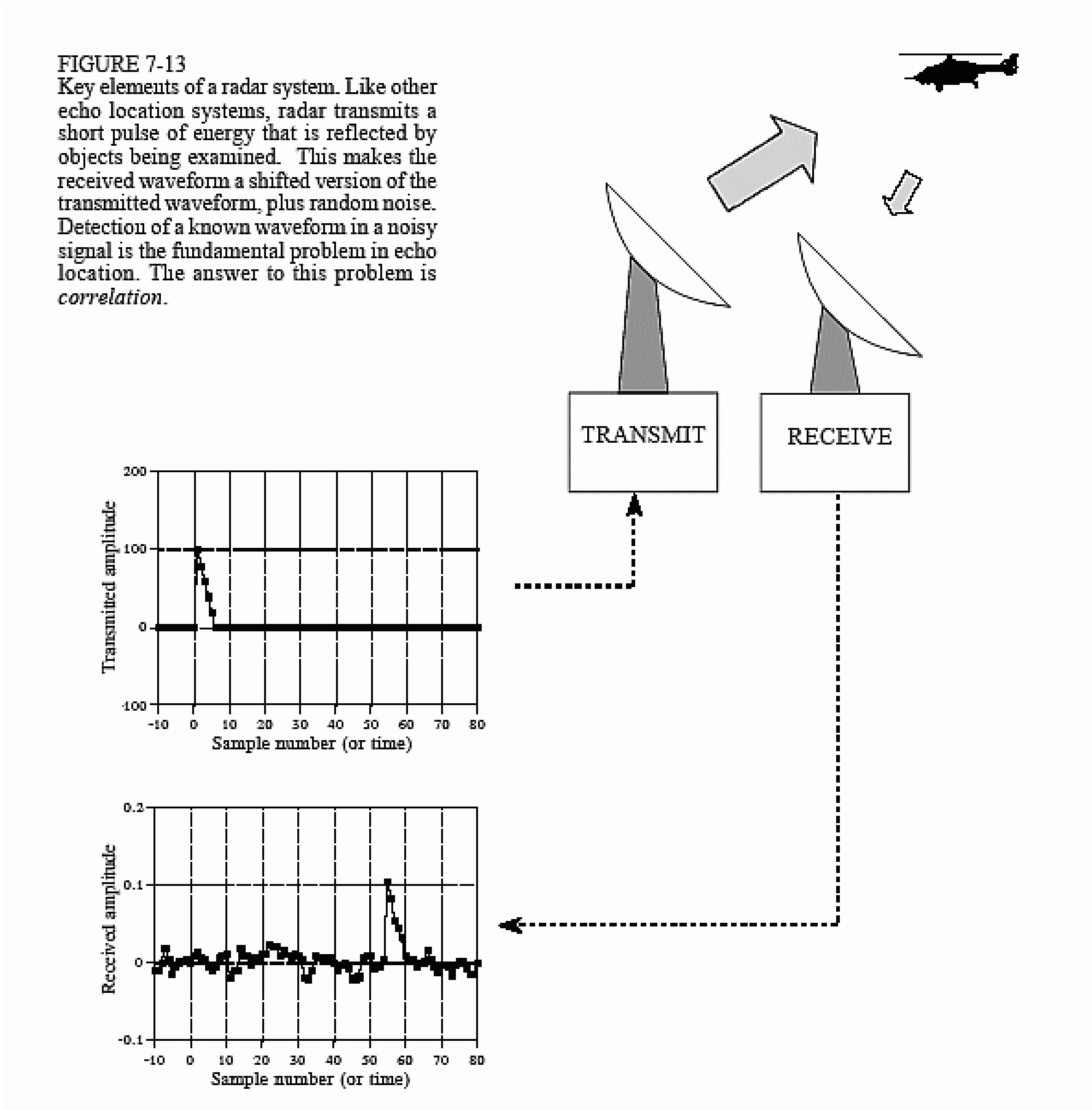


FIGURE 7-13
Key elements of a radar system. Like other echo location systems, radar transmits a short pulse of energy that is reflected by objects being examined. This makes the received waveform a shifted version of the transmitted waveform, plus random noise. Detection of a known waveform in a noisy signal is the fundamental problem in echo location. The answer to this problem is *correlation*.

The amplitude of each sample in the cross-correlation signal is a measure of how much the received signal *resembles* the target signal, *at that location*. This means that a peak will occur in the cross-correlation signal for every target signal that is present in the received signal. In other words, the value of the cross-correlation is maximized when the target signal is *aligned* with the same features in the received signal.

What if the target signal contains samples with a negative value? Nothing changes. Imagine that the correlation machine is positioned such that the target signal is perfectly aligned with the matching waveform in the received signal. As samples from the received signal fall into the correlation machine, they are multiplied by their matching samples in the target signal. Neglecting noise, a positive sample will be multiplied by itself, resulting in a positive number. Likewise, a negative sample will be multiplied by itself, also resulting in a positive number. Even if the target signal is completely negative, the peak in the cross-correlation will still be positive.

If there is noise on the received signal, there will also be noise on the cross-correlation signal. It is an unavoidable fact that random noise looks a certain amount like any target signal you can choose. The noise on the cross-correlation signal is simply measuring this similarity. Except for this noise, the peak generated in the cross-correlation signal is symmetrical between its left and right. This is true even if the target signal isn't symmetrical. In addition, the width of the peak is twice the width of the target signal. Remember, the cross-correlation is trying to *detect* the target signal, not *recreate* it. There is no reason to expect that the peak will even look like the target signal.

Correlation is the *optimal* technique for detecting a known waveform in random noise. That is, the peak is higher above the noise using correlation than can be produced by any other linear system. (To be perfectly correct, it is only optimal for *random white noise*). Using correlation to detect a known waveform is frequently called **matched filtering**.

The correlation machine and convolution machine are identical, except for one small difference. As discussed in the last chapter, the signal inside of the convolution machine is *flipped* left-for-right. This means that samples numbers: 1, 2, 3 … run from the right to the left. In the correlation machine this flip doesn't take place, and the samples run in the normal direction.

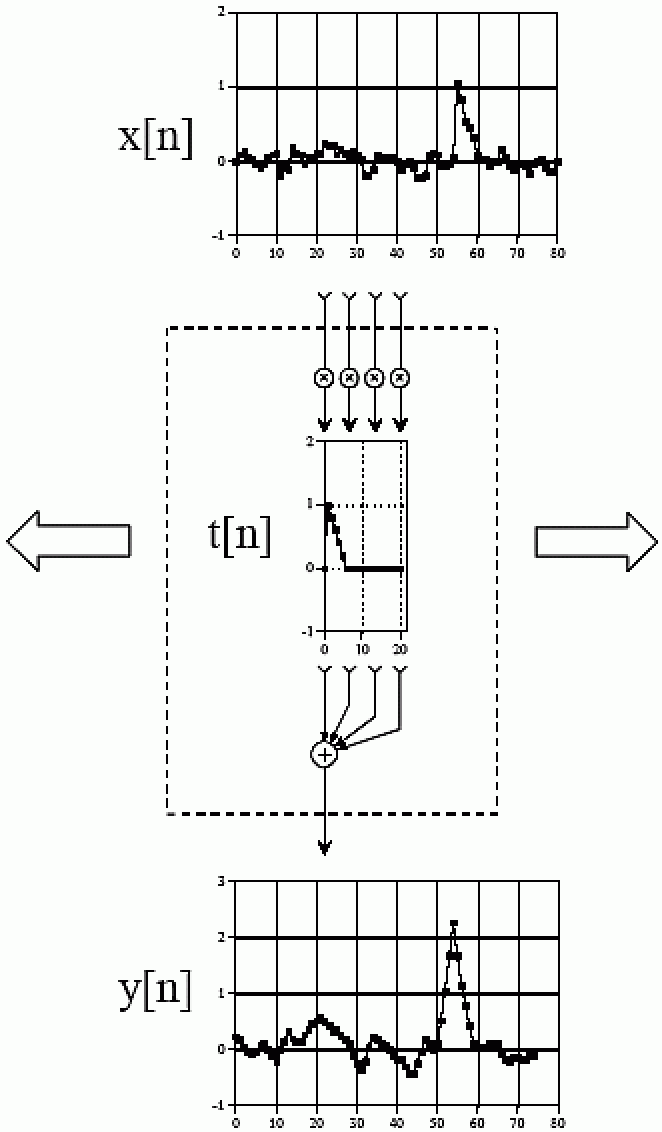


FIGURE 7-14
The correlation machine. This is a flowchart showing how the cross-correlation of two signals is calculated. In this example, $y[n]$ is the cross-correlation of $x[n]$ and $t[n]$. The dashed box is moved left or right so that its output points at the sample being calculated in $y[n]$. The indicated samples from $x[n]$ are multiplied by the corresponding samples in $t[n]$, and the products added. The correlation machine is identical to the convolution machine (Figs. 6-8 and 6-9), except that the signal inside of the dashed box is *not* reversed. In this illustration, the only samples calculated in $y[n]$ are where $t[n]$ is fully *immersed* in $x[n]$.

## Geortzel algorithm

The **Goertzel algorithm** is a digital signal processing (DSP) technique for identifying frequency components of a signal, published by Gerald Goertzel in 1958. While the general Fast Fourier transform (FFT) algorithm computes evenly across the bandwidth of the incoming signal, the Goertzel algorithm looks at specific, predetermined frequencies.

A practical application of this algorithm is recognition of the DTMF tones produced by the buttons pushed on a telephone keypad

It can also be used "in reverse" as a sinusoid synthesis function, which requires only 1 multiplication and 1 subtraction per sample.

### Explanation of algorithm

The Goertzel algorithm computes a sequence, s(n), given an input sequence, x(n):

$$s(n) = x(n) + 2\cos(2\pi\omega)s(n-1) - s(n-2)$$

where $s(-2) = s(-1) = 0$ and $\omega$ is some frequency of interest, in cycles per sample, which should be less than 1/2. This effectively implements a second-order IIR filter with poles at $e^{+2\pi i\omega}$ and $e^{-2\pi i\omega}$, and requires only one multiplication (assuming $2\cos(2\pi\omega)$ is pre-computed), one addition and one subtraction per input sample. For real inputs, these operations are real.

The Z transform of this process is

$$\frac{S(z)}{X(z)} = \frac{1}{1-2\cos(2\pi\omega)z^{-1}+z^{-2}} = \frac{1}{(1-e^{+2\pi i\omega}z^{-1})(1-e^{-2\pi i\omega}z^{-1})}$$

Applying an additional, FIR, transform of the form

$$\frac{Y(z)}{S(z)} = 1-e^{-2\pi i\omega}z^{-1}$$

will give an overall transform of

$$\frac{S(z)}{X(z)}\frac{Y(z)}{S(z)} = \frac{Y(z)}{X(z)} = \frac{(1-e^{-2\pi i\omega}z^{-1})}{(1-e^{+2\pi i\omega}z^{-1})(1-e^{-2\pi i\omega}z^{-1})} = \frac{1}{1-e^{+2\pi i\omega}z^{-1}}$$

The time-domain equivalent of this overall transform is

$$y(n) = x(n)+e^{+2\pi i\omega}y(n-1) = \sum_{k=-\infty}^{n} x(k)e^{+2\pi i\omega(n-k)} = e^{+2\pi i\omega n}\sum_{k=-\infty}^{n} x(k)e^{-2\pi i\omega k}$$

,

which becomes, assuming x(k) = 0 for all k < 0

$$y(n) = e^{+2\pi i\omega n} \sum_{k=0}^{n} x(k) e^{-2\pi i\omega k}$$

or, the equation for the (n + 1)-sample DFT of x, evaluated for ω and multiplied by the scale factor $e^{+2\pi i\omega n}$.

Note that applying the additional transform Y(z)/S(z) only requires the last two samples of the s sequence. Consequently, upon processing N samples x(0)...x(N − 1), the last two samples from the s sequence can be used to compute the value of a DFT bin, which corresponds to the chosen frequency ω as

$$X(\omega) = y(N-1)e^{-2\pi i\omega(N-1)} = (s(N-1) - e^{-2\pi i\omega}s(N-2))e^{-2\pi i\omega(N-1)}$$

For the special case often found when computing DFT bins, where ωN = k for some integer, k, this simplifies to

$$X(\omega) = (s(N-1) - e^{-2\pi i\omega}s(N-2))e^{+2\pi i\omega} = e^{+2\pi i\omega}s(N-1) - s(N-2)$$

In either case, the corresponding power can be computed using the same cosine term required to compute s as

$$X(\omega)X'(\omega) = s(N-2)^2 + s(N-1)^2 - 2\cos(2\pi\omega)s(N-2)s(N-1$$

## Least-Squares Linear-Phase FIR Filter Design

Let the FIR filter length be **L+1** samples, with $L$ even, and suppose we'll initially design it to be centered about the time origin. Then the frequency response is given on our frequency grid $\omega_k$ by

$$H(\omega_k) = \sum_{n=-L/2}^{L/2} h_n e^{-j\omega_k n}$$

Enforcing *even symmetry* in the impulse response, *i.e.*, $h_n = h_{-n}$, gives a *zero phase* FIR filter which we can later right-shift $L/2$ samples to make a causal, *linear phase* filter. In this case, the frequency response reduces to a *sum of cosines*:

$$H(\omega_k) \quad = \quad h_0 + 2\sum_{n=1}^{L/2} h_n \cos(\omega_k n), \quad k = 0, 1, 2, \ldots, N-1$$

or in matrix form:

$$\begin{bmatrix} H(\omega_0) \\ H(\omega_1) \\ \vdots \\ H(\omega_{N-1}) \end{bmatrix} = \underbrace{\begin{bmatrix} 1 & 2\cos(\omega_0) & \ldots & 2\cos[\omega_0(L-1)] \\ 1 & 2\cos(\omega_1) & \ldots & 2\cos[\omega_1(L-1)] \\ \vdots & & & \\ 1 & 2\cos(\omega_{N-1}) & \ldots & 2\cos[\omega_{N-1}(L-1)] \end{bmatrix}}_{A} \underbrace{\begin{bmatrix} h_0 \\ h_1 \\ \vdots \\ h_{L/2} \end{bmatrix}}_{x}$$

(Note that Remez exchange algorithms are also based on this formulation internally.)

**Matrix Formulation: Optimal $L2$Design, Cont'd**

In matrix notation, our filter design problem can be stated

$$\min_x \|Ax - b\|_2^2$$

where

$$A \triangleq \begin{bmatrix} 1 & 2\cos(\omega_0) & \ldots & 2\cos\left[\omega_0(L-1)\right] \\ 1 & 2\cos(\omega_1) & \ldots & 2\cos\left[\omega_1(L-1)\right] \\ \vdots & & & \\ 1 & 2\cos(\omega_{N-1}) & \ldots & 2\cos\left[\omega_{N-1}(L-1)\right] \end{bmatrix}$$

$$x \triangleq h$$

and $b = [D(\omega_k)]$ is the desired frequency response at the specified frequencies.

**Least Squares Optimization**

$$\hat{x} \stackrel{\Delta}{=} \arg\min_{x} \| Ax - b \|_2 = \arg\min_{x} \| Ax - b \|_2^2$$

Hence we can minimize

$$\| Ax - b \|_2^2 = (Ax - b)^T (Ax - b)$$

Expanding this, we have:

$$(Ax - b)^T (Ax - b) = (b^T - x^T A^T)(Ax - b)$$

This is quadratic in $x$, hence it has a *global minimum* which we can find by taking the derivative, setting it to zero, and solving for $x$. Doing this yields:

$$A^T Ax - A^T b = 0$$

These are the famous *normal equations* whose solution is given by:

$$\boxed{\hat{x} = \left[ (A^T A)^{-1} A^T \right] b}$$

The matrix

$$A^{\dagger} \stackrel{\Delta}{=} (A^T A)^{-1} A^T$$

is known as the (Moore-Penrose) *pseudo-inverse* of the matrix $A$.

**Geometrical Interpretation of Least Squares**

Typically, the number of frequency constraints is much greater than the number of design variables (filter taps). In these cases, we have an *overdetermined* system of equations (more equations than unknowns). Therefore, we cannot generally satisfy all the equations, and we are left with minimizing some error criterion to find the ``optimal compromise" solution.

In the case of least-squares approximation, we are minimizing the *Euclidean distance*, which suggests the geometrical interpretation shown in Fig.4.28.

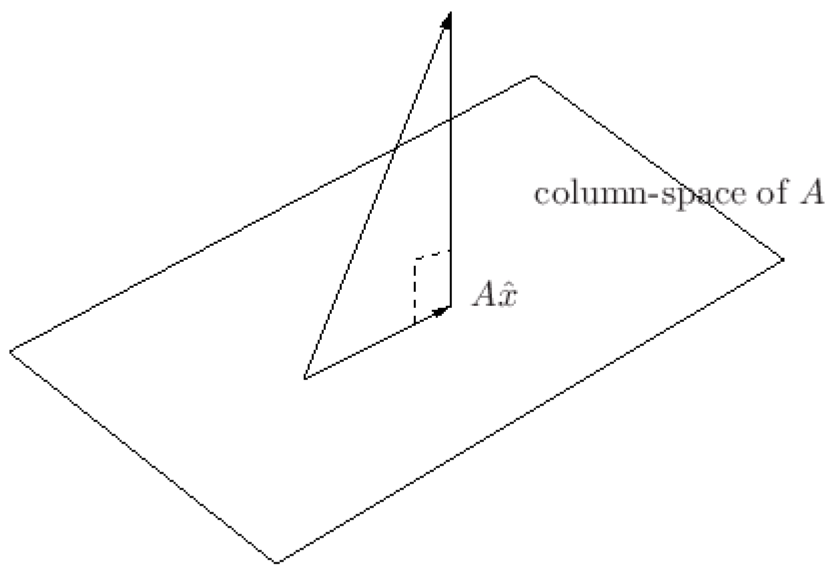


Figure 10.1: Geometrical interpretation of orthogonal projection.

Thus, the desired vector $b$ is the vector sum of its best least-squares approximation $A\hat{x}$ plus an orthogonal error $e$:

$$b = A\hat{x} + e.$$

In practice, the least-squares solution $\hat{x}$ can be found by minimizing the sum of squared errors:

$$\text{Minimize}_x \|e\|_2 = \|b - Ax\|_2$$

Figure 4.28 suggests that the error vector $b - A\hat{x}$ is *orthogonal* to the column space of the matrix $A$, hence it must be orthogonal to each column in $A$:

$$A^T(b - A\hat{x}) = 0 \Rightarrow A^T A\hat{x} = A^T b$$

This is how the *orthogonality principle* can be used to derive the fact that the best least squares solution is given by

$$\hat{x} = (A^T A)^{-1} A^T b = A^\dagger b$$

Note that the pseudo-inverse $A^\dagger$ *projects* the vector $b$ onto the column space of $A$.

(Note: To obtain the best numerical algorithms for least-squares solution in Matlab, it is usually better to use ``x = A \ b" rather than explicitly computing the pseudo-inverse as in ``x = pinv(A) * b".)

**4. Sampling Rate Conversion by Stages**

The decimator and interpolator discussed so far are of a single-stage structure. When large changes in sampling rate are required, multiple stages of sample rate conversion are found

to be more computationally efficient. Most practical systems employ a multi-stage structure, resulting in a considerable relaxation in the specifications of anti-aliasing (decimation)

or anti-imaging (interpolation) filters in each stage compared to a single stage realization. The decimation in Figure 3.23 can be realized in two stages if the decimation factor

$D$ can be expressed as a product of two integers, $D_1$ and $D_2$. Referring to Figure 3.24, in the first stage, the signal $x(n)$ is decimated by a factor of $D1$. The output, $v(p)$ is further decimated by $D_2$ in the second stage resulting in an overall decimation of $x(n)$ by

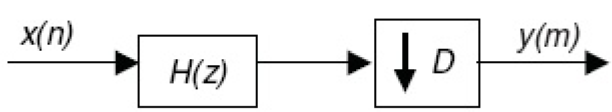


Figure 3.23: Decimation in a Single Stage.

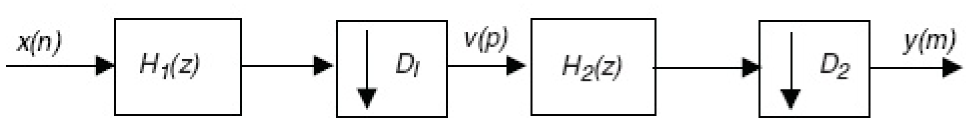


Figure 3.24: Decimation in Two Stages.

$D = (D_1 D_2)$. The filters $H_1(z)$ and $H_2(z)$ are so designed that the aliasing in the band of interest is belowa prescribed level and that the overall passband and stopband tolerances are met. This multi-stage sampling rate conversion system offers less computation and more flexibility in filter design. An example is given below to illustrate the idea of multi-stage sampling rate conversion.

**Example: Multi-Stage Sampling Rate Conversion**

We have a discrete time signal with a sampling rate of 90 kHz. The signal has the desired information in the frequency band from 0 to 450 Hz (passband), and the band from 450 to 500 Hz is the transition band. The signal is to be decimated by a factor of ninety. The required tolerances are a passband ripple of 0.002 and a stopband ripple of 0.001.

*Decimation in a Single Stage*

First we consider a single-stage design as shown in Figure 3.25(a). The specifications of the required LPF are shown in Figure 3.25(b).

According to the formula by Kaiser, the approximate length of an FIR filter is given by

$$N = \frac{-20 log_{10}\sqrt{\delta_p \delta_s} - 13}{14.6\Delta f} \qquad (3.37)$$

where peak passband ripple (linear) $\delta_p = 0.002$, peak stopband ripple (linear) $\delta_s = 0.001$, normalized transition bandwidth $\Delta f = \frac{f_s - f_p}{F_s}$, passband edge frequency $fp$ = 450 Hz, stopband edge frequency $fs$ = 500 Hz, and sampling frequency $F_s$ = 90 kHz.

From Equation 3.37, the lowpass FIR filter $H(z)$ has a length of $N \approx 5424$. Therefore, the number of multiplications per second, $M_{sec}$, needed for this single-stage decimator is

$$M_{sec} = 5424 * \frac{90000}{90} = 5,424,000. \qquad (3.38)$$

Since only one out of ninety samples is actually used, the computation rate is based on the decimated signal rate.

*Decimation in Two Stages*

Let us now consider the two-stage implementation of the decimation process as shown in Figure 3.26.

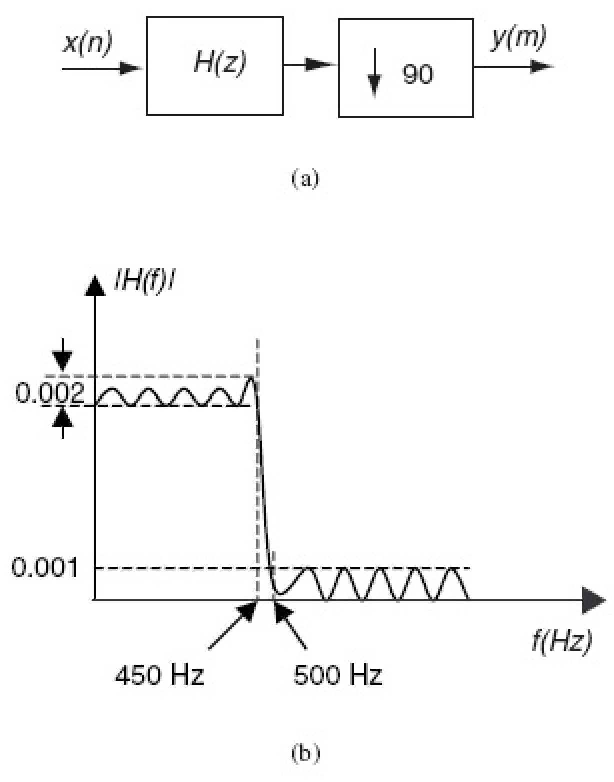


Figure 3.25: (a) Block Diagram for Single-Stage Decimation, (b) The Filter Specification.

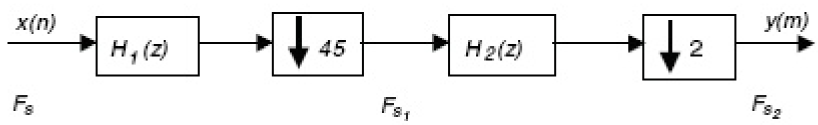


Figure 3.26: Block Diagram for Multi-Stage Decimation.

Due to the cascade decomposition, each of the two filters, $H_1(z)$ and $H_2(z)$, must have a linear passband ripple specification half of that specified for the single-stage filter, $H(z)$. The stopband ripple specifications for these two filters can be the same as that of $H(z)$ since the cascade connection will only reduce the stopband ripple.

**Stage One**

The first stage will decimate the input signal $x(n)$ by a factor of forty-five. The filter specifications for the first-stage LPF $H_1(z)$ are

$$\begin{aligned}\delta_{p_1} &= 0.001 = (0.002/2),\\ \delta_{s_1} &= 0.001,\\ f_{p_1} &= 450\ \text{Hz},\\ f_{s_1} &= 2000 - 500 = 1500\ \text{Hz},\end{aligned}$$

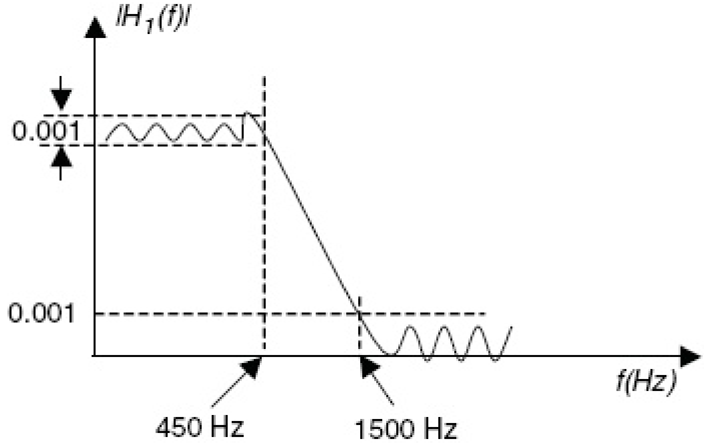


Figure 3.27: Decimation Filter Design for Stage One.

$$\begin{aligned}F_s &= 90\ \text{kHz, and}\\ F_{s_1} &= \frac{90{,}000}{45} = 2{,}000\ \text{Hz}.\end{aligned}$$

These specifications are shown in Figure 3.27.

The reason for choosing this value of the stopband edge is that, after decimation by a factor of forty-five, the residual energy of the signal in the band from 1000 to 2000 Hz will be aliased back to the band from 0 to 1000 Hz. Due to the attenuation in the stopband, the energy of the signal in the band from 1500 to 2000 Hz is very small compared to that in 1000 to 1500 Hz. So the amount of aliasing in the desired band of interest (0 to 450 Hz) will also be small, resulting in very little signal distortion.

According to Equation 3.37, the approximate length of the FIR filter, $H_1(z)$ is $N_1 =$ 276. The number of multiplications per second for the first stage is

$$M_{1,sec} = N_1 \frac{F_s}{D_1} = 276 * \frac{90000}{45} = 552,000. \quad (3.39)$$

**Stage Two**

The specifications for the second-stage filter, $H_2(z)$, are

$$\begin{aligned} \delta_{p_1} &= 0.001 = (0.002/2), \\ \delta_{s_1} &= 0.001, \\ f_{p_2} &= 450 \text{ Hz}, \\ f_{s_2} &= 500 \text{ Hz, and} \\ F_{s_1} &= 2000 \text{ Hz.} \end{aligned}$$

Figure 3.28 shows the characteristics of $H_2(z)$. This stage will perform a decimation of factor two on the output signal of the first stage. So, the total decimation of $x(n)$ is by a factor of ninety as required.

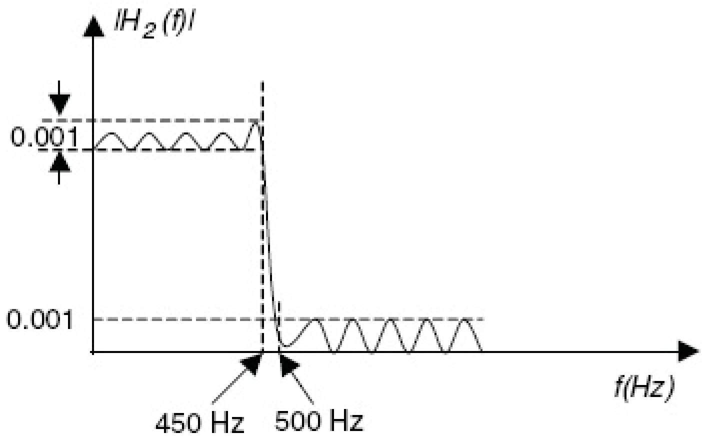


Figure 3.28: Decimation Filter Design for Stage Two.

For the second stage, the length of the filter, as calculated from Equation 3.37, is $N2$ = 129. The number of multiplications required for this stage is

$$M_{2,sec} = 129 * \frac{2000}{2} = 129,000. \quad (3.40)$$

The total number of multiplications per second required for the two-stage implementation of the decimator is

$$M_{sec} = M_{1,sec} + M_{2,sec} = 552,000 + 129,000 = 681,000. \qquad (3.41)$$

So, the two-stage implementation requires only $\frac{681,000}{5,424,000} = \frac{1}{8}$ of the operation required of the single-stage implementation

*Decimation in Three Stages*

To further illustrate the concept of multi-stage implementation of decimator and interpolator, we will now consider the three-stage implementation as shown in Figure 3.29.

**Stage One**

In this stage, decimation by fifteen is performed on the input signal $x(n)$. The characteristics of the LPF, $H_1(z)$, are shown in Figure 3.30. The filter specifications are

$$\begin{aligned}
\delta_{p1} &= 0.00067 = (0.002/3), \\
\delta_{s1} &= 0.001, \\
f_{p1} &= 450 \text{ Hz}, \\
f_{s1} &= 6000 - 500 = 5500 \text{ Hz}, \\
F_s &= 90 \text{ kHz, and} \\
F_{s1} &= \frac{90,000}{15} = 6,000 \text{ Hz.}
\end{aligned}$$

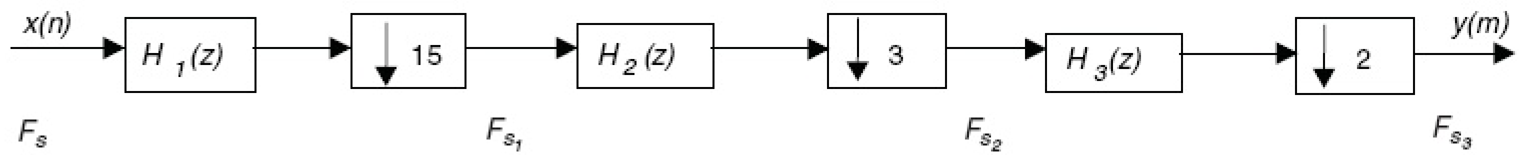


Figure 3.29: Block Diagram for Decimation in Three Stages.

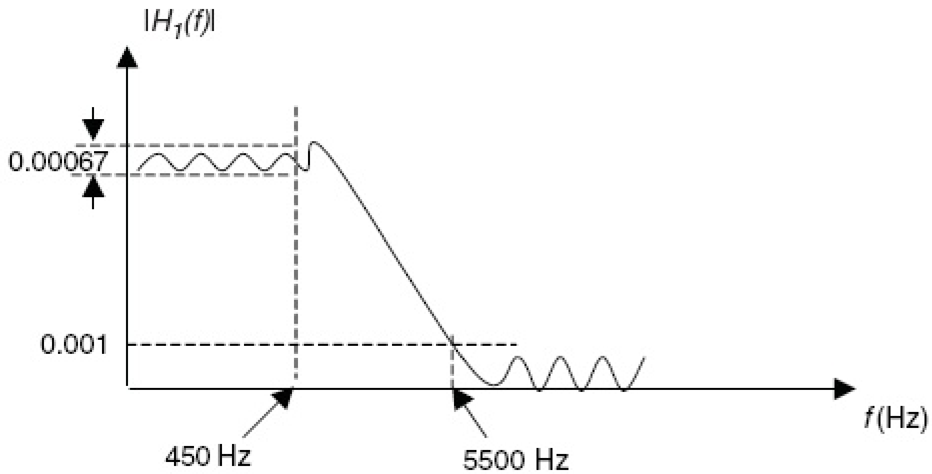


Figure 3.30: Decimation Filter Design for Stage One.

As in the two-stage case, the choice of stopband edge frequency can be extended to the point for which negligible aliasing occurs in the passband (band of interest).

The approximate length of the filter as given by Equation 3.37 is $N_1 = 60$. The number of multiplications per second for this stage is calculated as

$$M_{1,sec} = N_1 \frac{F_s}{D_1} = 60 * \frac{90000}{15} = 36,000. \qquad (3.42)$$

**Stage Two**

In this stage, a decimation by a factor of three is done. The specifications of the LPF in this stage, $H_2(z)$, are

$$\begin{aligned}
\delta_{p_2} &= 0.00067 = (0.002/3), \\
\delta_{s_2} &= 0.001, \\
f_{p_2} &= 450 \text{ Hz}, \\
f_{s_2} &= 2000 - 500 = 1500 \text{ Hz}, \\
F_{s_1} &= 6000 \text{ Hz, and} \\
F_{s_2} &= \tfrac{6000}{3} = 2000 \text{ Hz}.
\end{aligned}$$

As before, the stopband edge frequency can be stretched out to 1500 Hz. The filter characteristics are shown in Figure 3.31.

The length of filter required for this stage is $N_2 = 20$ and the number of multiplications per second is

$$M_{2,sec} = N_2 \frac{F_{s_1}}{D_2} = 20 * \frac{6000}{3} = 40,000. \qquad (3.43)$$

**Stage Three**

The third stage performs a decimation of factor two on the output of the second stage. The specifications of the LPF, $H_3(z)$, in this stage are

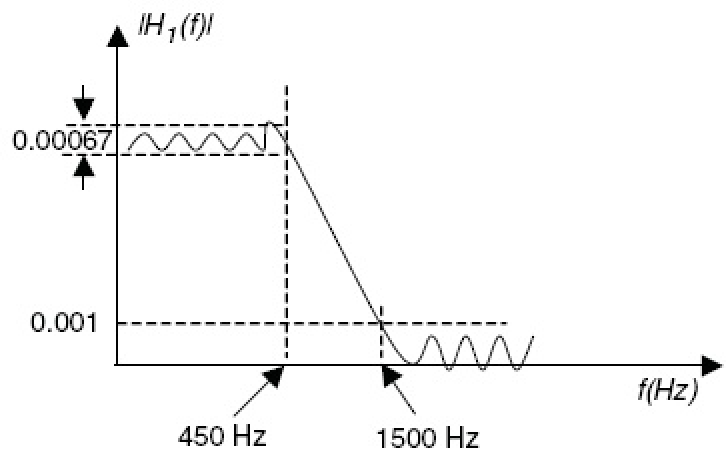


Figure 3.31: Decimation Filter Design for Stage Two.

$$\begin{aligned}
\delta_{p_3} &= 0.00067 = (0.002/3),\\
\delta_{s_3} &= 0.001,\\
f_{p3} &= 450 \text{ Hz},\\
f_{s3} &= 500 \text{ Hz},\\
F_{s_2} &= 2000 \text{ Hz, and}\\
F_{s_3} &= N_3 \frac{F_{s_2}}{D_3}.
\end{aligned}$$

Figure 3.32 shows the specifications for $H_3(z)$. As before, the approximate length of the filter, as calculated from Equation 3.37, is $N_3 = 134$. The number of multiplications required per second in the third stage is

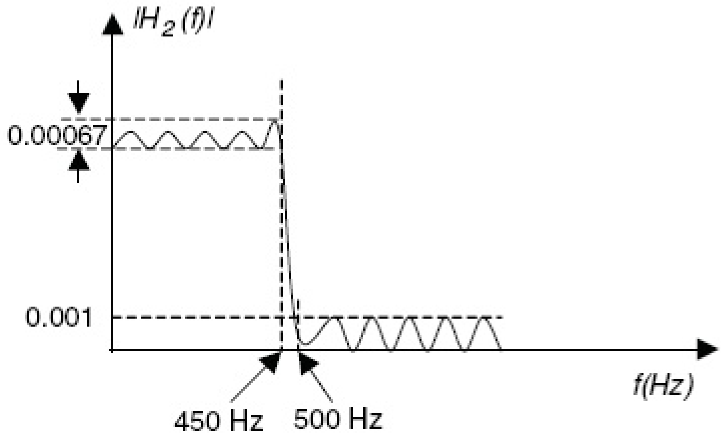


Figure 3.32: Decimation Filter Design for Stage Three.

$$M_{3,sec} = 134 * \frac{2000}{2} = 134,000. \qquad (3.44)$$

The total number of multiplications in the three stages of implementation is

$$M_{sec} = M_{1,sec} + M_{2,sec} + M_{3,sec} = 360,000 + 40,000 + 134,000 = 534,000. \quad (3.45)$$

[+] Enlarge Image

Compared to the single-stage implementation, the number ofmultiplicationsper second

are reduced by a factor of $\frac{5,424,000}{534,000} = 10$ by using three stages.

From this example, we can see that a significant saving in computation as well as in storage can be achieved by a multi-stage decimator and interpolator design. These savings depend on the optimum design of the number of stages and the choice of decimation factor for the individual stages.

The examples illustrate the many different combinations and ordering possible. One approach is to determine the sets of *I* and *D* factors that satisfy the filtering requirements and then estimate the storage and computational costs for each set. The lowest cost solution is then selected.

## 16. UNIVERSITY QUESTON PAPERS OF PRUVIOIUS YEARS

**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**
**DIGITAL SIGNAL PROCESSING**
**(COMMON TO EEE, ECE, EIE, ETM, ICE)**
**Time: 3hours Max. Marks: 80**
**Answer any FIVE questions**
**All Questions Carry Equal Marks**

1.a) Define an LTI System and show that the output of an LTI system is given by the convolution of Input sequence and impulse response.

b) Prove that the system defined by the following difference equation is an LTI system $y(n) = x(n+1)-3x(n)+x(n-1)$ ; $n\geq 0$. [8+8 ]

2.a) Define DFT and IDFT. State any Four properties of DFT.

b) Find 8-Point DFT of the given time domain sequence x(n) = {1, 2, 3, 4}. [8+8]

3.a) Derive the expressions for computing the FFT using DIT algorithm and hence draw the standard butterfly structure.

b) Compare the computational complexity of FFT and DFT. [8+8]

4. Discuss and draw various IIR realization structures like Direct form – I, Direct form-II, Parallel and cascade forms for the difference equation given $y(n) = - 3/8\, Y(n-1) + 3/32\, y(n-2) + 1/64\, y(n-3) + x(n) + 3\, x(n-1) + 2\, x(n-2)$.

5.a) Compare Butterworth and Chebyshev approximation techniques.

b) Design a Digital Butterworth LPF using Bilinear transformation technique for the following specifications

$0.707 \leq |H(w)| \leq 1$ ; $0 \leq w \leq 0.2\pi$

$|H(w)| \leq 0.08$ ; $0.4\pi \leq w \leq$ [ 8+8]

6.a) Compare FIR and IIR filters

b) Design an FIR Digital High pass filter using Hamming window whose cut off freq is 1.2 rad/s and length of window N=9. [8+8]

7.a) Define Multirate systems and Sampling rate conversion

b) Discuss the process of n Decimation by a factor D and explain how the aliasing effect can be eliminated. [8+8]

8. Discuss various Modified Bus structures of Programmable DSP Processors.[16]

**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**

**DIGITAL SIGNAL PROCESSING**

**(COMMON TO EEE, ECE, EIE, ETM, ICE)**

**Time: 3hours Max. Marks: 80**

**Answer any FIVE questions**

**All Questions Carry Equal Marks**

1.a) Write short notes on classification of systems.

b) Derive BIBO stability criteria to achieve stability of a system. [8+8]

2.a) Define DFS. State any Four properties of DFS.

b) Find the IDFT of the given sequence x(K) = {2, 2-3j, 2+3j, -2}. [8+8]

3.a) Find X(K) of the given sequence x(n) = { 1,2,3,4,4,3,2,1}using DIT- FFT algorithm.

b) Compare the computational complexity of FFT and DFT. [8+8]

4. What are the various basic building blocks in realization of Digital Systems and hence discuss transposed form realization structures.

5.a) Compare Impulse Invariant and Bilinear transformation techniques.

b) Compute the poles of an Analog Chebyshev filter TF that satisfies the Constraints

$$0.707 \le |H(j\Omega)| \le 1 \quad ; 0 \le \Omega \le 2$$
$$|H(j\Omega)| \le 0.1 \; ; \Omega \ge 4$$

and determine $H_a(s)$ and hence obtain H(z) using Bilinear transformation. [16]

6.a) Derive the conditions to achieve Linear Phase characteristics of FIR filters

b) Design an FIR Digital Low pass filter using Hanning window whose cut off freq is 2 rad/s and length of window N=9. [8+8]

7.a) Discuss the implementation of Polyphase filters for Interpolators with an example b) Discuss the sampling rate conversion by a factor I/D with the help of a Neat block Diagram. [8+8]

8. Write short notes on:
a) VLIW Architecture of Programmable Digital Signal Processors
b) Multiplier and Multiplier Accumulator [8+8]

**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**

**DIGITAL SIGNAL PROCESSING**

**(COMMON TO EEE, ECE, EIE, ETM, ICE)**

**Time: 3hours Max. Marks: 80**

**Answer any FIVE questions**

**All Questions Carry Equal Marks**

1.a) Discuss various discrete time sequences.
b) Give the Basic block diagram of Digital Signal Processor. [8+8]

2.a) Define DFS. State any Four properties of DFS.
b) Find the IDFT of the given sequence x(K) = {2, 2-3j, 2+3j, -2}. [8+8]

3.a) Find IFFT of the given X(K) = { 1,2,3,4,4,3,2,1}using DIF algorithm
b) Bring out the relationship between DFT and Z-transform. [8+8]

4.a) Define Z-Transform and List out its properties.
b) Discuss Direct form, Cascade and Linear phase realization structures of FIR filters. [8+8]

5.a) Discuss digital and analog frequency transformation techniques.
b) Discuss IIR filter design using Bilinear transformation and hence discuss frequency warping effect. [8+8]

6.a) Compare various windowing functions.
b) Design an FIR Digital Low pass filter using rectangular window whose cut off freq is 2 rad/s and length of window N=9. [8+8]

7.a) Define Interpolation and Decimation. List out the advantages of Sampling rate conversion.

b) Discuss the sampling rate conversion by a factor I with the help of a Neat block Diagram.[8+8]

8.a) Discuss Various Addressing modes of Programmable Digital Signal Processors. b) Give the Internal Architecture of TMS320C5X 16 bit fixed point processor.[ 8+8]

**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**

**DIGITAL SIGNAL PROCESSING**

**(COMMON TO EEE, ECE, EIE, ETM, ICE)**

**Time: 3hours Max. Marks: 80**

**Answer any FIVE questions**

**All Questions Carry Equal Marks**

1.a) Define Linearity, Time Invariant, Stability and Causality.

b) The discrete time system is represented by the following difference equations in which x(n) is input and y(n) is output. $Y(n) = 3y^2(n-1) - nx(n)+4x(n-1)-2x(n-1)$.

[8+8]

2.a) Define Convolution. Compare Linear and Circular Convolution techniques. b) Find the Linear convolution of the given two sequences x(n)={1,2} and h(n) ={1,2,3} using DFT and IDFT.

[8+8]

3.a) Develop DIT-FFT algorithm and draw signal flow graphs for decomposing the DFT for N=6 by considering the factors for N = 6 = 2.3.

b) Bring out the relationship between DFT and Z-transform. [8+8]

4.a) Discuss transposed form structures with an example.

b) Discuss Direct form, Cascade realization structures of FIR filters. [8+8]

5.a) Discuss digital and analog frequency transformation techniques.

b) Discuss IIR filter design using Impulse Invariant transformation and list out its advantages and Limitations. [8+8]

6.a) Compare various windowing functions

b) Design an FIR Digital Band pass filter using rectangular window whose upper and lower cut off freq.'s are 1 & 2 rad/s and length of window N = 9. [8+8]

7.a) Define Interpolation and Decimation.

b) Discuss the sampling rate conversion by a factor I/D with the help of a Neat block Diagram. [8 +8]

8.a) Write a short notes on On-Chip peripherals of Programmable DSP's.

b)Give the Internal Architecture of TMS320C5X 16 bit fixed point processor. [8+8]



**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**

**DIGITAL SIGNAL PROCESSING**

**(COMMON TO EEE, ECE, EIE, ETM, ICE)**

**Time: 3hours Max. Marks: 80**

**Answer any FIVE questions**

**All Questions Carry Equal Marks**

1. (a) Discuss impulse invariance method of deriving IIR digital filter from corre- sponding analog filter.

   (b) Use the Bilinear transformation to convert the analog filter with system func- tion $H(S) = S + 0.1/(S + 0.1)^2 + 9$ into a digital IIR filters. Select $T = 0.1$ and compare the location of the zeros in H(Z) with the locations of the zeros obtained by applying the impulse invariance method in the conversion of H(S). [8+8]

2. (a) Design a high pass filter using hamming window with a cut-off frequency of 1.2 radians/second and N=9

   (b) Compare FIR and IIR filters. [10+6]

3. (a) For each of the following systems, determine whether or not the system is i. stable

   ii. causal

   iii. linear

   iv. shift-invariant.

   A. $T[x(n)] = x(n - n_0)$ B. $T[x(n)] = e^{x}(n)$

C. $T[x(n)] = a\,x(n) + b$.
Justify your answer.

(b) A system is described by the difference equation y(n)-y(n-1)-y(n-2) = x(n-1). Assuming that the system is initially relaxed, determine its unit sample response h(n). [8+8]

4. (a) Implement the decimation in time FFT algorithm for N=16.

(b) In the above Question how many non - trivial multiplications are Required.

5. (a) Discuss the frequency-domain representation of discrete-time systems and sig- nals by deriving the necessary relation.

(b) Draw the frequency response of LSI system with impulse response $h(n) = a^n u(-n)\ (|a| < 1)$

6. (a) Describe how targets can be decided using RADAR

(b) Give an expression for the following parameters relative to RADAR

i. Beam width
ii. Maximum unambiguous range

(c) Discuss signal processing in a RADAR system. [6+6+4]

7. (a) An LTI system is described by the equation y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2). Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane.

(b) Define stable and unstable systems. Test the condition for stability of the first-order IIR filter governed by the equation y(n)=x(n)+bx(n-1). [8+8]

8. (a) Compute Discrete Fourier transform of the following finite length sequence considered to be of length N.

i. $x(n) = \delta(n + n_0)$ where $0 < n_0 < N$
ii. $x(n) = a^n$ where $0 < a < 1$.

(b) If x(n) denotes a finite length sequence of length N, show that $x((-n))_N = x((N - n))_N$. [8+8]

**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**
**DIGITAL SIGNAL PROCESSING**
**(COMMON TO EEE, ECE, EIE, ETM, ICE)**
**Time: 3hours Max. Marks: 80**
**Answer any FIVE questions**
**All Questions Carry Equal Marks**

1. (a) An LTI system is described by the equation y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2). Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane.

   (b) Define stable and unstable systems. Test the condition for stability of the first-order IIR filter governed by the equation y(n)=x(n)+bx(n-1). [8+8]

2. (a) Discuss impulse invariance method of deriving IIR digital filter from corre- sponding analog filter.

   (b) Use the Bilinear transformation to convert the analog filter with system func- tion $H(S) = S + 0.1/(S + 0.1)^2 + 9$ into a digital IIR filters. Select $T = 0.1$ and compare the location of the zeros in H(Z) with the locations of the zeros obtained by applying the impulse invariance method in the conversion of H(S). [8+8]

3. (a) Describe how targets can be decided using RADAR

   (b) Give an expression for the following parameters relative to RADAR

      i. Beam width
      ii. Maximum unambiguous range

   (c) Discuss signal processing in a RADAR system. [6+6+4]

4. (a) Discuss the frequency-domain representation of discrete-time systems and sig- nals by deriving the necessary relation.

   (b) Draw the frequency response of LSI system with impulse response $h(n) = a^n u(-n)$ $(|a| < 1)$ [8+8]

5. (a) For each of the following systems, determine whether or not the system is i. stable

      ii. causal
      iii. linear
      iv. shift-invariant.

   A. $T[x(n)] = x(n - n_0)$ B. $T[x(n)] = e^x(n)$

C. $T[x(n)] = a$
$x(n) + b$.
Justify your
answer.

(b) A system is described by the difference equation y(n)-y(n-1)-y(n-2) = x(n-
1). Assuming that the system is initially relaxed, determine its unit sample response h(n). [8+8]

6. (a) Implement the decimation in time FFT algorithm for N=16.

(b) In the above Question how many non - trivial multiplications are Required.

7. (a) Design a high pass filter using hamming window with a cut-off frequency of
1.2 radians/second and N=9

(b) Compare FIR and IIR filters.

[10+6]

8. (a) Compute Discrete Fourier transform of the following finite length sequence considered to be of length N.

i. $x(n) = \delta(n + n_0)$ where $0 < n_0 < N$

ii. $x(n) = a^n$ where $0 < a < 1$.

(b) If x(n) denotes a finite length sequence of length N, show that $x((-n))_N = x((N - n))_N$.[8+8]



**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**
**DIGITAL SIGNAL PROCESSING**
**(COMMON TO EEE, ECE, EIE, ETM, ICE)**
**Time: 3hours Max. Marks: 80**
**Answer any FIVE questions**
**All Questions Carry Equal Marks**

1. (a) Design a high pass filter using hamming window with a cut-off frequency of
1.2 radians/second and N=9

(b) Compare FIR and IIR filters.

[10+6]

2. (a) Describe how targets can be decided using RADAR

(b) Give an expression for the following parameters relative to RADAR

i. Beam width

ii. Maximum unambiguous range

(c) Discuss signal processing in a RADAR system.

[6 +6+4]

3. (a) An LTI system is described by the equation y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2). Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane.

   (b) Define stable and unstable systems. Test the condition for stability of the first-order IIR filter governed by the equation y(n)=x(n)+bx(n-1). [8+8]

4. (a) Compute Discrete Fourier transform of the following finite length sequence considered to be of length N.

   i. $x(n) = \delta(n + n_0)$ where $0 < n_0 < N$
   ii. $x(n) = a^n$ where $0 < a < 1$.

   (b) If x(n) denotes a finite length sequence of length N, show that $x((-n))_N = x((N - n))_N$. [8+8]

5. (a) For each of the following systems, determine whether or not the system is i. stable

   ii. causal
   iii. linear
   iv. shift-invariant.

   A. $T[x(n)] = x(n - n_0)$ B. $T[x(n)] = e^{x}(n)$

   C. $T[x(n)] = a\,x(n) + b$.

Justify your answer

   (b) A system is described by the difference equation y(n)-y(n-1)-y(n-2) = x(n-1). Assuming that the system is initially relaxed, determine its unit sample response h(n). [8+8]

6. (a) Discuss the frequency-domain representation of discrete-time systems and sig- nals by deriving the necessary relation.

   (b) Draw the frequency response of LSI system with impulse response $h(n) = a^n u(-n)$ $(|a| < 1)$ [8+8]

7. (a) Implement the decimation in time FFT algorithm for N=16.

(b) In the above Question how many non - trivial multiplications are Required.

8. (a) Discuss impulse invariance method of deriving IIR digital filter from corre- sponding analog filter.

(b) Use the Bilinear transformation to convert the analog filter with system func- tion $H(S) = S + 0.1/(S + 0.1)^2 + 9$ into a digital IIR filters. Select $T = 0.1$ and compare the location of the zeros in H(Z) with the locations of the zeros obtained by applying the impulse invariance method in the conversion of H(S).
[8+8

**III B.TECH - II SEMESTER EXAMINATIONS, APRIL/MAY, 2011**
**DIGITAL SIGNAL PROCESSING**
**(COMMON TO EEE, ECE, EIE, ETM, ICE)**
**Time: 3hours Max. Marks: 80**
**Answer any FIVE questions**
**All Questions Carry Equal Marks**

1. (a) Compute Discrete Fourier transform of the following finite length sequence considered to be of length N.

   i. $x(n) = \delta(n + n_0)$ where $0 < n_0 < N$
   ii. $x(n) = a^n$ where $0 < a < 1$.

   (b) If x(n) denotes a finite length sequence of length N, show that $x((-n))_N = x((N - n))_N$. [8+8]

2. (a) For each of the following systems, determine whether or not the system is i. stable
   ii. causal
   iii. linear
   iv. shift-invariant.

   A. $T[x(n)] = x(n - n_0)$ B. $T[x(n)] = e^{x}(n)$
   C. $T[x(n)] = a\, x(n) + b$. Justify your answer.

   (b) A system is described by the difference equation y(n)-y(n-1)-y(n-2) = x(n-1). Assuming that the system is initially relaxed, determine its unit sample response h(n). [8+8]

[6+6+4]

4. (a) Design a high pass filter using hamming window with a cut-off frequency of
1.2 radians/second and N=9
(b) Compare FIR and IIR filters.

[10+6]

5. (a) An LTI system is described by the equation y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2). Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane
(b) Define stable and unstable systems. Test the condition for stability of the first-order IIR filter governed by the equation y(n)=x(n)+bx(n-1). [8+8]

6. (a) Discuss the frequency-domain representation of discrete-time systems and sig- nals by deriving the necessary relation.
(b) Draw the frequency response of LSI system with impulse response
$h(n) = a^n u(-n)$ $(|a| < 1)$ [8+8]

7. (a) Discuss impulse invariance method of deriving IIR digital filter from corre- sponding analog filter.
(b) Use the Bilinear transformation to convert the analog filter with system function $H(S) = S + 0.1/(S + 0.1)^2 + 9$ into a digital IIR filters. Select $T = 0.1$ and compare the location of the zeros in H(Z)with the locations of the zeros obtained by applying the impulse invariance method in the conversion of H(S). [8+8]

8. (a) Implement the decimation in time FFT algorithm forN=16.
(b) In the above Question how many non - trivial Multiplications are required.

**17. <u>QUESTION BANK</u>:**

**PART - A (10 x 2 = 20 Marks)**

1. What is the system impulse response if the input and output are $x(n)=(1/2)^n u(n)$, $y(n)=(1/2)^n u(n)$ respectively?
2. Determine the circular convolution of the sequence $x_1(n)=\{1,2,3,1\}$, $x_2(n)=\{4,3,2,2\}$
3. What are the advantages and disadvantages of FIR over IIR filter?
4. Convert the non-recursive system $H(z) = 1 + z^{-1} + z^{-2} + z^{-3} + z^{-4}$ into recursive system.
5. How are the limit cycle oscillations due to overflow minimized?
6. Determine the direct form realizations for the filter $h(n)=\{1,2,3,4,3,2,1\}$
7. What is the effect of product Quantization due to finite word length?
8. What are the advantages of multistage implementation in multirate signal processing?
9. Define periodogram? How can it be smoothed?
10. Where will you place zero & poles in a filter to eliminate 50 Hz frequency in a sampled signal at sampling frequency F=600Hz?

**PART - B (5 x 16 = 80 Marks)**

11. Using FFT algorithm compute the output of linear filter described by $h(n)=\{1,2,3,2,1\}$ and input $x(n)=\{1,1,1,1\}$

12.a) Design a Chebyshev digital low pass filter with the following specifications. pass band ripple <= 1 dB, pass band edge = 4Khz,stop band attenuation >= 40 dB, stop band edge = 6Khz & sampling rate = 24Khz. Use bilinear transformation.

**(OR)**

12.b) Design a Butterworth IIR filter with the following specifications

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

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

Use Impulse Invariant method.

13.a) Design an FIR linear phase digital filter approximating the ideal frequency response,

$H_d(\omega) = 1, \quad 0 \le |\omega| \le \pi/6$

$0, \quad \pi/6 < |\omega| \le \pi$

Determine the coefficients of a 13-tap filter based on window method. Use hamming window.

**(OR)**

13.b) Design an FIR digital filter whose frequency spectral samples are

$H(k) = e^{-j16\pi k/17} \quad 0 \le k \le 4$

$= 0 \quad 5 \le k \le 12$

$= e^{-j16\pi(k-17)/17} \quad 13 \le k \le 16$

Use Hanning window.

14.a) Consider the system $y(n) = 0.8575\, y(n-1) - 0.125\, y(n-2) + x(n)$

i) Compute the poles & Design the cascade realizations of the system.

ii) Quantize the coefficients of the system using truncation, maintains a sign bit plus three other bits. Determine the poles of the resulting system.

iii) Determine the resulting frequency at -3dB. Assume sampling frequency as $F_s$= 1000Hz.

**(OR)**

14.b) The transfer function for an FIR filter is given by $H(z)=1-1.334335z^{-1} + 0.9025z^{-1}$ Draw the realization diagram for each of the following cases. (i) Transversal structures (ii) a two-stage lattice structure. Calculate the values of the coefficients for the lattice structure.

15.a) Consider the signal x $(n)=a^n u(n)$, $|a|<1$.Determine the spectrum $X(\omega)$.The signal x(n) is applied to a decimator that reduces the rate by a factor of 2. Determine the output spectrum. Discuss the design criteria for anti-aliasing filter.
15.b) Determine the Power Spectral density estimate of the signal $x(n)=(0.9)^n$,0 <= n <= 20 using Blackman-Tukey method.

TWO MARKS:
1. Define about DFT and IDFT?
2. Find the values of WN
k ,When N=8, k=2 and also for k=3.
3. Compare DIT radix-2 FFT and DIF radix -2 FFT.
4. Draw the radix-2 FFT–DIF butterfly diagram.
5. Draw the radix-2 FFT–DIT butterfly diagram.
6. What is the necessity of sectioned convolution in signal processing?
7. Define Correlation of the sequence.
8. State any two DFT properties.
9. Why impulse invariant transformation is not a one-to-one mapping?
PART - B
1. a) Compute 4- point DFT of casual three sample sequence is given by,
x(n) = 1/3, 0_n_2
= 0, else (10)
b) State and prove shifting property of DFT. (6)
2. Derive and draw the radix -2 DIT algorithms for FFT of 8 points. (16)
3. Compute the DFT for the sequence {1, 2, 0, 0, 0, 2, 1, 1}. Using radix -2 DIF FFT and
radix -2 DIT- FFT algorithm. (16)
4. Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal
x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}. Using Overlap add overlap save method. (16)
5. In an LTI system the input x(n) = {1, 1, 1}and the impulse response h(n) = {-1, -}Determine
the response of LTI system by radix -2 DIT FFT (16)
6. Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal
x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}. Using Overlap save method (16)

TWO MARKS:
1. Differentiate IIR filters and FIR filters.
2. Write the characteristics features of Hanning window
3. Define pre-warping effect? Why it is employed?
4. Give any two properties of Butterworth filter.
5. When a FIR filter is said to be a linear phase FIR filter
6. Write the characteristics features of rectangular window.
7. Write the expression for Kaiser window function..

8. What are the advantages and disadvantages of FIR filters?
9. Write the characteristics features of Hamming window
10. Why mapping is needed in the design of digital filters?

PART - B

1 . With a neat sketch explain the design of IIR filter using impulse invariant transformation. (16)
2. Apply impulse invariant transformation to H(S) =
(S +1) (S + 2)
with T =1sec and find H(Z). (16)
3. For a given specifications of the desired low pass filter is
0.707 _ |H(_)| _1.0, 0 _ _ _ 0.2_
|H(_)| _ 0.08, 0.4 _ _ _ _ _
Design a Butterworth filter using bilinear transformation. (16)
4. Explain the procedural steps the design of low pass digital Butterworth filter and list its
properties. (16)
5. The normalized transfer function of an analog filter is given by,
1
Ha(Sn)= Sn
2 + 1.414Sn +1
with a cutoff frequency of 0.4 _, using bilinear transformation. (16)
.
6. List the three well known methods of design technique for IIR filters and explain any one.(16)
7. Design a low pass filter using rectangular window by taking 9 samples of w(n) and with a cutoff frequency of 1.2 radians/sec.
Using frequency sampling method, design a band pass FIR filter with the following specification. Sampling frequency Fs =8000 Hz, Cutoff frequency fc1 =1000Hz, fc2 =3000Hz.Determine the filter coefficients for N =7. (16)
8. Design an ideal high pass filter with Hd(ej _) = 1 ; _/4 _ | _| _ _
= 0 ; | _| _ _/4 Using Hamming window with N =11 (16)
9. Determine the coefficients of a linear phase FIR filter of length N =15 which has a symmetric
unit sample response and a frequency response that satisfies the conditions
H (2 _k /15) = 1 ; for k = 0, 1, 2, 3
0.4 ; for k = 4
0 ; for k = 5, 6, 7 (16)
10. Design and implement linear phase FIR filter of length N =15 which has following unit sample
sequence H(k) = 1 ; for k = 0, 1, 2, 3
0 ; for k =4, 5, 6, 7 (16)
11. Convert the analog filter in to a digital filter whose system function is
S + 0.2 H(s)= ---------------- .Use Impulse Invariant Transformation .Assume T=1sec (16)

(S + 0.2)2 + 9
1
12. The Analog Transfer function H(s)= ----------------.Determine H(Z) .Using Impulse (S+1) (S+2) Invariant Transformation .Assume T=1sec . (8)
13. Apply Bilinear Transformation to H(s)= ------------- with T=0.1 sec. (8)
(S+2)(S+3)

TWO MARKS:
1. What are the effects of finite word length in digital filters?
2. List the errors which arise due to quantization process.
3. Discuss the truncation error in quantization process.
4. Write expression for variance of round-off quantization noise.
5. What is sampling?
6. Define limit cycle Oscillations, and list out the types.
7. When zero limit cycle oscillation and Over flow limit cycle oscillation has occur?
8. Why? Scaling is important in Finite word length effect.
9. What are the differences between Fixed and Binary floating point number representation?
10. What is the error range for Truncation and round-off process?

PART - B
1. The output of an A/D is fed through a digital system whose system function is H(Z)=(1-_)z /(z-_), 0<_<1.Find the output noise power of the digital system. (8)
2. The output of an A/D is fed through a digital system whose system function is H(Z)=0.6z/z-0.6. Find the output noise power of the digital system=8 bits (8)
3. Discuss in detail about quantization effect in ADC of signals. Derive the expression for Pe(n) and SNR. (16)
4 a. Write a short notes on limit cycle oscillation (8)
b. Explain in detail about signal scaling (8)
5. A digital system is characterized by the difference equation Y(n)=0.95y(n-1)+x(n).determine the dead band of the system when x(n)=0 and y(-1)=13. (16)
6. Two first order filters are connected in cascaded whose system functions of theIndividual sections are H1(z)=1/(1-0.8$z^{-1}$ ) and H2(z)=1/(1-0.9$z^{-1}$ ).Determine theOver all output noise power. (16)

TWO MARKS QUESTIONS:
1. What is the need for spectral estimation?
2. How can the energy density spectrum be determined?
3. What is autocorrelation function?
4. What is the relationship between autocorrelation and spectral density?
5. Give the estimate of autocorrelation function and power density for random signals?
6. Obtain the expression for mean and variance for the autocorrelation function of random

signals.
7. Define period gram.
PART - B
1. Explain how DFT and FFT are useful in power spectral estimation.(10)
2. Explain Power spectrum estimation using the Bartlett window.(8)
3. Obtain the mean and variance of the averaging modified period gram estimate.(16)
4. How is the Blackman and Tukey method used in smoothing the periodogram?(10)
5. Derive the mean and variance of the power spectral estimate of the Blackman and Tukey
method.(10)
6. What are the limitations of non-parametric methods in spectral estimation?(8)
7. How the parametric methods overcome the limitations of the non-parametric methods?(10)
TWO MARKS:
1. What are the factors that influence the selection of DSPs.
2 .What are the advantages and disadvantages of VLIW architecture?
3. What is pipelining? and What are the stages of pipelining?
4. What are the different buses of TMS 320C5x processor and list their functions
5. List the various registers used with ARAU.
6. What are the shift instructions in TMS 320 C5x.
7. List the on-chip peripherals of C5x processor.
PART - B
1. Explain in detail about the applications of PDSP (10)
2. Explain briefly:
(i). Von Neumann architecture (5)
(ii). Harvard architecture (5)
(iii).VLIW architecture (6)
3. Explain in detail about
(i). MAC unit (8 )
(ii). Pipelining (8)
4. Draw and explain the architecture of TMS 320C5x processor (16)
5. Explain in detail about the Addressing modes of TMS 320C50 (16)

**18. <u>ASSIGNMENT TOPICS:</u>**

## Assignment Questions
## UNIT-I

1.a) Define an LTI System and show that the output of an LTI system is given by the convolution of Input sequence and impulse response.
b) Prove that the system defined by the following difference onequati is an LTI system $y(n) = x(n+1)-3x(n)+x(n-1)$ ; $n \geq 0$.

2.a) Write short notes on classification of systems.

b) Derive BIBO stability criteria to achieve stability of a system.

3.a) Discuss various discrete time sequences.

b) Give the Basic block diagram of Digital Signal Processor.

4.a) Define Linearity, Time Invariant, Stability and Causality.

b) The discrete time system is represented by the following difference equations in which x(n) is input and y(n) is output. $Y(n) = 3y^2(n-1)-nx(n)+4x(n-1)-2x(n-1)$.

5. (a) Discuss impulse invariance method of deriving IIR digital filter from corresponding analog filter.

(b) Use the Bilinear transformation to convert the analog filter with system func- tion $H(S) = S + 0.1/(S + 0.1)^2 + 9$ into a digital IIR filters. Select $T = 0.1$ and compare the location of the zeros in H(Z) with the locations of the zeros obtained by applying the impulse invariance method in the conversion of H(S).

6.a) Define Z-Transform and List out its properties.

b) Discuss Direct form, Cascade and Linear phase realization structures of FIR filters.

7.a) Discuss transposed form structures with an example.

b) Discuss Direct form, Cascade realization structures of FIR filters.

8. Discuss and draw various IIR realization structures like Direct form – I, Direct form-II, Parallel and cascade forms for the difference equation given by

$$y(n) = - 3/8\, Y(n-1) + 3/32\, y(n-2) + 1/64\, y(n-3) + x(n) + 3\, x(n-1)$$

9. What are the various basic building blocks in realization of Digital Systems and hence discuss transposed form realization structures.

(a) Implement the decimation in time FFT algorithm for N=16.

(b) In the above Question how many non - trivial multiplications are Required.

## UNIT-II

1.a) Define DFS. State any Four properties of DFS.

b) Find the IDFT of the given sequence x(K) = {2, 2-3j, 2+3j, -2}.

2.a) Define Convolution. Compare Linear and Circular Convolution techniques. b) Find the Linear convolution of the given two

sequences x(n)={1,2} and
h(n) ={1,2,3} using DFT and IDFT.

3.(a) Design a high pass filter using hamming window with a cut-off frequency 1.2 radians/second and N=9

(b) Compare FIR and IIR filters.

4.a) Define DFS. State any Four properties of DFS.

b) Find the IDFT of the given sequence x(K) = {2, 2-3j, 2+3j, -2}.

5.a) Define DFT and IDFT. State any Four properties of DFT.

b) Find 8-Point DFT of the given time domain sequence x(n) = {1, 2, 3, 4}.

5.a) Find IFFT of the given X(K) = { 1,2,3,4,4,3,2,1}using DIF algorithm

b) Bring out the relationship between DFT and Z-transform.

6.a) Develop DIT-FFT algorithm and draw signal flow graphs for decomposing the DFT for N=6 by considering the factors for N = 6 = 2.3.

b) Bring out the relationship between DFT and Z-transform.

7. (a) For each of the following systems, determine whether or not the system is i. stable

ii. causal

iii. linear

iv. shift-invariant.

A. $T[x(n)] = x(n - n_0)$ B. $T[x(n)] = e^{x}(n)$

C. $T[x(n)] = a\,x(n) + b$. Justify your answer.

(b) A system is described by the difference equation y(n)-y(n-1)-y(n-2) = x(n-1). Assuming that the system is initially relaxed, determine its unit sample response h(n).

8.a) Derive the expressions for computing the FFT using DIT algorithm and hence draw the standard butterfly structure.

b) Compare the computational complexity of FFT and DFT.

3.a) Find X(K) of the given sequence x(n) = { 1,2,3,4,4,3,2,1}using DIT- FFT algorithm.

b) Compare the computational complexity of FFT and DFT.

## UNIT-IV

1.a) Discuss digital and analog frequency transformation techniques.
b) Discuss IIR filter design using Bilinear transformation and hence discuss frequency warping effect.

2.a) Discuss digital and analog frequency transformation techniques.
b) Discuss IIR filter design using Impulse Invariant transformation and list out its advantages and Limitations.

3. (a) Discuss the frequency-domain representation of discrete-time systems and sig- nals by deriving the necessary relation.

(b) Draw the frequency response of LSI system with impulse response $h(n) = a^n u(-n)$ $(|a| < 1)$

4.a) Compare Butterworth and Chebyshev approximation techniques.
b) Design a Digital Butterworth LPF using Bilinear transformation technique for the following specifications

$0.707 \le |H(w)| \le 1$ ; $0 \le w \le 0.2\pi$

$|H(w)| \le 0.08$ ; $0.4\pi \le w \le \pi$

5.a) Compare Impulse Invariant and Bilinear transformation techniques.
b) Compute the poles of an Analog Chebyshev filter TF that satisfies the Constraints

$0.707 \le |H(j\Omega)| \le 1$ ; $0 \le \Omega \le 2$

$|H(j\Omega)| \le 0.1$ ; $\Omega \ge 4$

and determine $H_a(s)$ and hence obtain $H(z)$ using Bilinear transformation.

## UNIT-IV

1.a) Derive the conditions to achieve Linear Phase characteristics of FIR filters
b) Design an FIR Digital Low pass filter using Hanning window whose cut 1ff freq is 2 rad/s and length of window N=9.

2.a) Compare FIR and IIR filters
b) Design an FIR Digital High pass filter using Hamming window whose cut off freq is 1.2 rad/s and length of window N=9.
a) Compare various windowing functions
b) Design an FIR Digital Band pass filter using rectangular window whose upper and lower cut off freq.'s are 1 & 2 rad/s and length of

window N = 9.

3.a) Compare various windowing functions.

b) Design an FIR Digital Low pass filter using rectangular window whose cut off freq is 2 rad/s and length of window N=9.

4. (a) Describe how targets can be decided using RADAR

(b) Give an expression for the following parameters relative to RADAR

i. Beam width

ii. Maximum unambiguous range

(c) Discuss signal processing in a RADAR system.

## UNIT-V

1. a) Discuss the implementation of Polyphase filters for Interpolators with an example

b) Discuss the sampling rate conversion by a factor I/D with the help of a Neat block Diagram.

2. a) Define Interpolation and Decimation.

b) Discuss the sampling rate conversion by a factor I/D with the help of aNeat block Diagram.

3.a) Define Interpolation and Decimation. List out the advantages of Sampling rate conversion.

b) Discuss the sampling rate conversion by a factor I with the help of a Neat block Diagram.

4. a) Define Multirate systems and Sampling rate conversion

b) Discuss the process of n Decimation by a factor D and explain how the aliasing effect can be eliminated

5.(a) An LTI system is described by the equation
$y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2)$.
Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane.

(b) Define stable and unstable systems. Test the condition for stability of the first-order IIR filter governed by the equation $y(n)=x(n)+bx(n-1)$.

6.Discuss various Modified Bus structures of Programmable DSP Processors.

Write short notes on:

7. a) VLIW Architecture of Programmable Digital Signal Processors

b) Multiplier and Multiplier Accumulator

8. a) Discuss Various Addressing modes of Programmable Digital Signal Processors.

b) Give the Internal Architecture of TMS320C5X 16 bit fixed point processor

9. a) Write a short notes on On-Chip peripherals of Programmable DSP's.

b)Give the Internal Architecture of TMS320C5X 16 bit fixed point processor

10. (a) Compute Discrete Fourier transform of the following finite length sequence considered to

be of length N.

i. $x(n) = \delta(n + n_0)$ where $0 < n_0 < N$

ii. $x(n) = a^n$ where $0 < a < 1$.

(b) If x(n) denotes a finite length sequence of length N, show that $x((-n))_N = x((N - n))_N$



**19. UNIT WISE QUIZ QUESTIONS AND LONG ANSWER QUESTIONS:**

## UNIT-I: INTRODUCTION

1. Sketch the discrete time signal x(n) =4 _ (n+4) + _(n)+ 2 _ (n-1) + _ (n-2) -5 _ (n-3**)**.
2. What is the causality condition for an LTI system?
3. Define stable and unstable systems.
4. Test the condition for stability of the first-order IIR filter governed by the equation y(n)=x(n)+bx(n-1).
5. Define Linearity, Time Invariant, Stability and Causality.

**REALIZATION OF DIGITAL FILTERS**

1. What are two different types of structures for realization of IIR systems?

2. Explain the below.

1. Direct-form1 structure

2.Direct-form2 structure

3.Transposed direct-form2 structure

4.Cascade form structure

5.Parallel form structure

6.Lattice-ladder structure

3. Define signal flow graph.

4. A signal flow graph is a graphical representation of the relationship between the variables of a set of linear difference equations.

5. What is the advantage of cascade realization?

6. How Quantization errors can be minimized if we realize an LTI system in cascade form

**UNIT II: DISCRETE FOURIER SERIES**

1.Compare DFS,DTFT & DFT.
2. State and prove the properties of DFT.
3. Define DFS. State any Four properties of DFS.
4.Explain about overlap-add and overlap save method.

**FAST FOURIER TRANSFORMS**

1. Define about DFT and IDFT?

2. Find the values of $W_{N}^{k}$ ,When N=8, k=2 and also for k=3.

3. Compare DIT radix-2 FFT and DIF radix -2 FFT.

4. Draw the radix-2 FFT-DIF butterfly diagram.

5. Draw the radix-2 FFT-DIT butterfly diagram.

6. What is the necessity of sectioned convolution in signal processing?

7. Define Correlation of the sequence.

8. State any two DFT properties.

9. Why impulse invariant transformation is not a one-to-one mapping?

1. a) Compute 4- point DFT of casual three sample sequence is given by,

**x(n) = 1/3, 0_n_2 = 0, else**

b) State and prove shifting property of DFT.

2. Derive and draw the radix -2 DIT algorithms for FFT of 8 points.

3. Compute the DFT for the sequence {1, 2, 0, 0, 0, 2, 1, 1}. Using radix -2 DIF FFT and radix -2 DIT- FFT algorithm.

4. Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}. Using Overlap add overlap save method.

5. In an LTI system the input x(n) = {1, 1, 1}and the impulse response h(n) = {-1, -}Determine the response of LTI system by radix -2 DIT FFT

6. Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}. Using Overlap save method

## UNIT IV: IIR DIGITAL FILTERS

1 . With a neat sketch explain the design of IIR filter using impulse invariant transformation.

2. Apply impulse invariant transformation to **H(S) = (S +1) (S + 2)** with T =1sec and find H(Z).

3. For a given specifications of the desired low pass filter is

0.707 _ |H(_)| _1.0, 0 _ _ _ 0.2_|H(_)| _ 0.08, 0.4 _ _ _ _ _

Design a Butterworth filter using bilinear transformation.

4. Explain the procedural steps the design of low pass digital Butterworth filter and list its properties.

5. The normalized transfer function of an analog filter is given by,

1$H_a(S_n)= S_n^2 + 1.414S_n +1$ with a cutoff frequency of 0.4 _, using bilinear transformation.

6. List the three well known methods of design technique for IIR filters and explain any one.

7. Design a low pass filter using rectangular window by taking 9 samples of w(n) and with a cutoff frequency of 1.2 radians/sec.Using frequency sampling method, design a band pass FIR filter with the following specification. Sampling frequency $F_s$ =8000 Hz, Cutoff frequency $fc_1$ =1000Hz, $fc_2$=3000Hz.Determine the filter coefficients for N =7.

8. Design an ideal high pass filter with $H_d(e^{j\_})$ = 1 ; _/4 _ | _| _ _= 0 ; | _| _ _/4 Using Hamming window with N =11

9. Determine the coefficients of a linear phase FIR filter of length N =15 which has a symmetric unit sample response and a frequency response that satisfies the conditions H (2 _k /15) = 1; for k = 0, 1, 2, 3

= 0.4 ; for k = 4

= 0; for k = 5, 6, 7

10. Design and implement linear phase FIR filter of length N =15 which has following unit sample sequence H(k) = 1 ; for k = 0, 1, 2, 3

= 0 ; for k =4, 5, 6, 7

11. Convert the analog filter in to a digital filter whose system function is

S + 0.2 H(s) = $(S + 0.2)^2$ + 9.Use Impulse Invariant Transformation .Assume T=1sec

12. The Analog Transfer function H(s)= ----------------.Determine H(Z) .Using Impulse (S+1)(S+2) Invariant Transformation .Assume T=1sec .

13. Apply Bilinear Transformation to H(s)= ------------- with T=0.1 sec.

(S+2)(S+3)

**UNIT V: FIR DIGITAL FILTERS**

1. Differentiate IIR filters and FIR filters.
2. Write the characteristics features of Hanning window
3. Define pre-warping effect? Why it is employed?
4. Give any two properties of Butterworth filter.
5. When a FIR filter is said to be a linear phase FIR filter
6. Write the characteristics features of rectangular window.
7. Write the expression for Kaiser window function..
8. What are the advantages and disadvantages of FIR filters?
9. Write the characteristics features of Hamming window
10. Why mapping is needed in the design of digital filters?

**UNIT V: MULTIRATE DIGITAL SIGNAL PROCESSING**

1. a) Discuss the implementation of Polyphase filters for Interpolators with an example
 b) Discuss the sampling rate conversion by a factor I/D with the help of a Neat block Diagram.
2. a) Define Interpolation and Decimation.
 b) Discuss the sampling rate conversion by a factor I/D with the help of aNeat block Diagram.

3.a) Define Interpolation and Decimation. List out the advantages of Sampling rate conversion.
 b) Discuss the sampling rate conversion by a factor I with the help of a Neat block Diagram.
4. a) Define Multirate systems and Sampling rate conversion
 b) Discuss the process of n Decimation by a factor D and explain how the aliasing effect can be eliminated
5.(a) An LTI system is described by the equation $y(n)=x(n)+0.81x(n-1)-0.81x(n-2)-0.45y(n-2)$.
Determine the transfer function of the system. Sketch the poles and zeroes on the Z-plane.

(b) Define stable and unstable systems. Test the condition for stability of the first-order IIR filter governed by the equation $y(n)=x(n)+bx(n-1)$.

## Finite word length effects in Digital filters:

1. What are the effects of finite word length in digital filters?

2. List the errors which arise due to quantization process.

3. Discuss the truncation error in quantization process.

4. Write expression for variance of round-off quantization noise.

5. What is sampling?

6. Define limit cycle Oscillations, and list out the types.

7. When zero limit cycle oscillation and Over flow limit cycle oscillation has occur?

8. Why? Scaling is important in Finite word length effect.

9. What are the differences between Fixed and Binary floating point number representation?

10. What is the error range for Truncation and round-off process?

**PART - B**

1. The output of an A/D is fed through a digital system whose system function is H(Z)=(1-_)z /(z-_), 0<_<1.Find the output noise power of the digital system.

2. The output of an A/D is fed through a digital system whose system function is H(Z)=0.6z/z-0.6. Find the output noise power of the digital system=8 bits

3. Discuss in detail about quantization effect in ADC of signals. Derive the expression for Pe(n) and SNR.

4 a. Write short notes on limit cycle oscillation

b. Explain in detail about signal scaling

5. A digital system is characterized by the difference equation Y(n)=0.95y(n-1)+x(n).determine the dead band of the system when x(n)=0 and y(-1)=13.

6. Two first order filters are connected in cascaded whose system functions of theIndividual sections are H1(z)=$1/(1-0.8z^{-1})$ and H2(z)=$1/(1-0.9z^{1})$.Determine the Over all output noise power.

## DIGITAL SIGNAL PROCESSORS

**TWO MARKS:**

1. What are the factors that influence the selection of DSPs.

2 .What are the advantages and disadvantages of VLIW architecture?

3. What is pipelining? and What are the stages of pipelining?

4. What are the different buses of TMS 320C5x processor and list their functions

5. List the various registers used with ARAU.

6. What are the shift instructions in TMS 320 C5x.

7. List the on-chip peripherals of C5x processor.

**PART - B**

1. Explain in detail about the applications of PDSP

2. Explain briefly:

(i). Von Neumann architecture

(ii). Harvard architecture

(iii).VLIW architecture

3. Explain in detail about

(i). MAC unit

(ii). Pipelining

4. Draw and explain the architecture of TMS 320C5x processor

5. Explain in detail about the Addressing modes of TMS 320C50

ASSIGNMENT QUESTIONS:

UNIT-1

1. Determine the energy of the discrete time sequence (2)
x(n) = (½)$_n$, n_0 =3 $_n$, n<0
2. Define multi channel and multi dimensional signals. (2)
3. Define symmetric and anti symmetric signals. (2)
4. Differentiate recursive and non recursive difference equations. (2)
5. What is meant by impulse response? (2)
6. What is meant by LTI system? (2)
7. What are the basic steps involved in convolution? (2)
8. Define the Auto correlation and Cross correlation? (2)
9. What is the causality condition for an LTI system? (2)
10. What are the different methods of evaluating inverse z transform? (2)
11. What is meant by ROC? (2)
12. What are the properties of ROC? (2)
13. What is zero padding? What are it uses? (2)
14. What is an anti imaging and anti aliasing filter? (2)
15. State the Sampling Theorem. (2)
16. Determine the signals are periodic and find the fundamental period (2)
i) sin_ 2 _t
ii) sin 20_t+ sin5_t
17. Give the mathematical and graphical representations of a unit sample, unit step sequence. (2)
18. Sketch the discrete time signal x(n) =4 _ (n+4) + _(n)+ 2 _ (n-1) + _ (n-2) -5 _ (n-3) (2)
19. Find the periodicity of x(n) =cost(2_n / 7) (2)
20. What is inverse system? (2)
21. Write the relationship between system function and the frequency response. (2)
22. Define commutative and associative law of convolutions. (2)
23. What is meant by Nyquist rate and Nyquist interval? (2)
24. What is an aliasing? How to overcome this effect? (2)
25. What are the disadvantages of DSP? (2)
26.Compare linear and circular convolution.(2)
27.What is meant by section convolution? (2)
28.Compare over lap add and save method. (2)
29. Define system function. (2)

30.State Parseval's relation in z - transform. (2)

## PART B

1. Determine whether the following system are linear, time-invariant (16)

(a)y(n) = Ax(n) +B. (4)

i(a)y(n) =x(2n). (4)

ii(a)y(n) =n x2 (n). (4)

iv)y(n) = a x(n) (4)

2. Check for following systems are linear, causal, time in variant, stable, static (16)

i) y(n) =x(2n). (4)

ii) y(n) = cos (x(n)). (4)

iii) y(n) = x(n) cos (x(n) (4)

iv) y(n) =x(-n+2) (4)

3. (a)For each impulse response determine the system is (a) stable i(a) causal

i) h(n)= sin **(**_ n / 2) . (4)

ii) h(n) = _(n) + sin _ n (4)

(b)Find the periodicity of the signal x(n) =sin **(2**_n / 3)+ **cos (**_ n / 2) (8)

4. (a)Find the periodicity of the signal

i) x(n) = **cos** (_ /4) **cos(**_n /4). (4)

ii) x(n) = cos (_ n 2 / 8) (4)

(b) State and proof of sampling theorem. (8)

5. Explain in detail about A to D conversion with suitable block diagram and to reconstruct the signal. (16)

6. What are the advantages of DSP over analog signal processing? (16)

**CONVOLUTION:**

**8.** Find the output of an LTI system if the input is x(n) =(n+2) for 0_ n_ 3

and h(n) =an u(n) for all n (16)

9. Find the convolution sum of x(n) =1 n = -2,0,1

= 2 n= -1

= 0 elsewhere

and h(n) = _ (n) – _ (n-1) + _( n-2) - _ (n-3). (16)

10. (a)Find the convolution of the following sequence x(n) = u(n) ; h(n) =u(n-3). (8)

(b)Find the convolution of the following sequence x(n) =(1,2,-1,1) , h(n) =(1, 0 ,1,1). (8 )
12.Find the output sequence y(n) if h(n) =(1,1,1) and x(n) =(1,2,3,1) using a circular Convolution. (16)
13. Find the convolution y(n) of the signals (16)
x(n) ={ _ $_n$, -3 _ n _ 5 and h(n) ={ 1, 0 _ n _ 4
0, elsewhere } 0, elsewhere }

**UN IT-II :**
20. The impulse response of LTI system is h(n)=(1,2,1,-1).Find the response of the system to the input x(n)=(2,1,0,2) (16)
21. Determine the magnitude and phase response of the given equation
y(n) =x(n)+x(n-2) (16)
22. Determine the response of the causal system y(n) – y(n-1) =x(n) + x(n-1) to inputs x(n)=u(n) and x(n) =2 $^{-n}$ u(n).Test its stability (16)
23. Determine the frequency response for the system given by
y(n)-y3/4y(n-1)+1/8 y(n-2) = x(n)- x(n-1) (16)
24. Determine the pole and zero plot for the system described difference equations
y(n)=x(n)+2x(n-1)-4x(n-2)+x(n-3) (16)
25.A system has unit sample response h(n) =-1/4 _(n+1)+1/2 _(n)-1/4 _(n-1).Is the system BIBO stable? Is the filter is Causal? Find the frequency response? (16)
26. Find the output of the system whose input- output is related by the difference equation
y(n) -5/6 y(n-1) +1/6 y(n-2) = x(n) -1/2 x(n-1) for the step input. (16)
27. Find the output of the system whose input- output is related by the difference equation
y(n) -5/6 y(n-1) +1/6 y(n-2) = x(n) -1/2 x(n-1) for the x(n) =4 $^n$ u(n). (16)

**UNIT – III**
**FAST FOURIER TRANSFORM**
**PART A ( 2marks)**
1. How many multiplication and additions are required to compute N point DFT using radix 2 FFT? (2)
2. Define DTFT pair. (2)
3. What are Twiddle factors of the DFT? (2)
4. State Periodicity Property of DFT. (2)

5. What is the difference between DFT and DTFT? (2)
6. Why need of FFT? (2)
7. Find the IDFT of Y (k) = (1, 0, 1, 0) (2)
8. Compute the Fourier transform of the signal x(n) = u(n) – u(n-1). (2)
9. Compare DIT and DIF? (2)
10. What is meant by in place in DIT and DIF algorithm? (2)
11. Is the DFT of a finite length sequence is periodic? If so, state the reason. (2)
12. Draw the butterfly operation in DIT and DIF algorithm? (2)
13. What is meant by radix 2 FFT? (2)
14. State the properties of $W_N^k$? (2)
15. What is bit reversal in FFT? (2)
16. Determine the no of bits required in computing the DFT of a 1024 point sequence with SNR of 30dB. (2)
17. What is the use of Fourier transform? (2)
18. What are the advantages FFT over DFT? (2)
19. What is meant by section convolution? (2)
20. Differentiate overlap adds and save method? (2)
21.Distinguish between Fourier series and Fourier transform. (2)
22.What is the relation between fourier transform and z transform. (2)
23. Distinguish between DFT and DTFT. (2)

**PART B**

1.(a) Determine the Fourier transform of $x(n) = a^{|n|}$; $-1<1$ (8)
(b) Determine the Inverse Fourier transform $H(w) = (1-ae^{-jw})^{-1}$ (8)
2. State and proof the properties of Fourier transform (16)

**FFT:**

3. Determine the Discrete Fourier transform x (n) = (1, 1, 1, 1) (16)
4. Derive and draw the 8 point FFT-DIT butterfly structure. (16)
5. Derive and draw the 8 point FFT-DIF butterfly structure. (16)
6.Compute the DFT for the sequence.(0.5,0.5,0.5,0.5,0,0,0,0) (16)

7.Compute the DFT for the sequence.(1,1,1,1,1,1,0,0) (16)

8.Find the DFT of a sequence x(n)=(1,1,0,0) and find IDFT of Y(k) =(1,0,1,0) (16)

9. If x (n) = sin (n_/2), n=0, 1, 2, 3

h (n) = $2^n$, n=0,1,2,3.Find IDFT and sketch it. (16)

10.(a) Find 4 point DFT using DIF of x(n) =(0,1,2,3) (8)

(b).Proof x(n)*h(n) =X(z) H(z) (8)

11.Discuss the properties of DFT. (16)

12.Discuss the use of FFT algorithm in linear filtering. (16)

13.Explain the application of DFT in linear filtering and spectral analysis? (16)

## UNIT –V IIR FILTER DESIGN

### PART A ( 2 marks)

1. Define canonic and non canonic form realizations**.** (2)
2. Draw the direct form realizations of FIR systems? (2)
3. Mention advantages of direct form II and cascade structure? (2)
4. Define Bilinear Transformation. (2)
5. What is prewar ping? Why is it needed? (2)
6. Write the expression for location of poles of normalized Butterworth filter. (2)
7. Distinguish between FIR and IIR Filters. (2)
8. What is linear phase filter? (2)
9. What are the design techniques available for IIR filter? (2)
10. What is the main drawback of impulse invariant mapping? (2)
11. Compare impulse invariant and bilinear transformation. (2)
12. Why IIR filters do not have linear phase? (2)
13. Mention the properties of Butterworth filter? (2)
14. Mention the properties of Chebyshev filter? (2)
15. Why impulse invariant method is not preferred in the design of high pass IIR filter? (2)
16. Give the transform relation for converting LPF to BPF in digital domain. (2)

### PART - B

**Structures of IIR systems:**

1. Obtain the cascade and parallel form realizations for the following systems (16)

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

2. Obtain the Direct form I and II

y (n) = -0.1(n-1) + 0.72 y(n-2) + 0.7x(n) -0.252 x(n-2) (16)

3. Obtain the (a) Direct forms i(a) cascade ii(a) parallel form realizations for the following systems y (n) = 3/4(n-1) – 1/8 y(n-2) + x(n) +1/3 x(n-1) (16)

4. Find the direct form I and II

H (z) =8z-2+5z-1+1 / 7z-3+8z-2+1 (16)

5.Find the direct form –I, cascade and parallel form for (16)

H(Z) = z -1 -1 / 1 – 0.5 z-1+0.06 z-2

**IIR FILTER DESIGN:**

6. Explain the method of design of IIR filters using bilinear transform method. (16)

7. (a)Discuss the limitations of designing an IIR filter using impulse invariant method. (8)

(b)Derive bilinear transformation for an analog filter with system function H(s) = b/ s + a (8)

8 (a)For the analog transfer function H(s) = 2 / (s+1) (s+3) .Determine H (z) using bilinear transformation. With T=0.1 sec (8)

(b)Convert the analog filter H(s) = 0.5 (s+4) / (s+1)(s+2) using impulse invariant transformation T=0.31416s (8)

9. The normalized transfer function of an analog filter is given by

$H_a(s_n) = 1/ s_n$

2 +1.414 s n +1. Convert analog filter to digital filter with cut off

frequency of 0.4 _ using bilinear transformation. (16)

10. Design a single pole low pass digital IIR filter with -3db bandwidth of 0.2_ by using bilinear transformation. (16)

11. For the constraints

0.8 _ |H (e jw)| _1, 0 _ _ _ 0.2_

|H (e jw)| _0.2, 0.6_ _ _ __ with T= 1 sec .Determine system

function H(z) for a Butterworth filter using Bilinear transformation. (16)

12.Design a digital Butterworth filter satisfying the following specifications

0.7 _ |H (e jw)| _1, 0 _ _ _ 0.2_

|H (e jw)| _0.2, 0.6_ _ _ __ with T= 1 sec .Determine system

function H(z) for a Butterworth filter using impulse invariant transformation. (16)

13. Design a digital Chebyshev low pass filter satisfying the following specifications

0.707 _ |H (e jw)| _1, 0 _ _ _ 0.2_

|H (e jw)| _0.1 0.5 _ _ __ with T= 1 sec using for bilinear

transformation. (16)

14.Design a digital Butterworth High pass filter satisfying the following specifications

0.9 _ |H (e jw)| _1, 0 _ _ _ _/2

|H (e jw)| _0.2, 3_/4 _ _ __ with T= 1 sec. using impulse

invariant transformation. (16)

15. Design a realize a digital filter using bilinear transformation for the following specifications

i) Monotonic pass band and stop band

ii) -3.01 db cut off at 0.5 _ rad

iii) Magnitude down at least 15 db at _ = 0.75 _ rad. (16)

**UNIT – VI FIR FILTER DESIGN**

**PART A ( 2marks)**

1. What are Gibbs oscillations?(2)
2. Explain briefly Hamming window(2).
3. If the impulse response of the symmetric linear phase FIR filter of length 5 is h(n) = {2, 3, 0, x, y), then find the values of x and y.(2)
4. What are the desirable properties of windowing technique?(2)
5. Write the equation of Bartlett window.(2)
6. Why IIR filters do not have linear phase?(2)
7. Why FIR filters are always stable?(2)
8. Why rectangular window are not used in FIR filter design using window method?(2)
9. What are the advantages of FIR filter? (2)
10. What are the advantages and disadvantages of window? (2)
11. What is the necessary condition and sufficient condition for the linear phase characteristic of a FIR filter? (2)
12. Compare Hamming and Hanning window? (2)
13. Why triangular window is not a good choice for designing FIR Filter? (2)
14. Why Kaiser window is most used for designing FIR Filter? (2)
15. What is the advantages in linear phase realization of FIR systems? (2)

**PART B**

1. Prove that an FIR filter has linear phase if the unit sample response satisfies the condition h(n) = ± h(M-1-n), n =0,1,….. M-1.Also discuss symmetric and anti symmetric cases of FIR filter. (16)
2. Explain the need for the use of window sequence in the design of FIR filter. Describe the

window sequence generally used and compare the properties. (16)

3. Explain the type 1 design of FIR filter using Frequency sampling technique. (16)

4. A LPF has the desired response given below (16)

H (e j ) = e -3 j , 0 _ _ _ _/2

0. _/2 _ _ _ _ .Determine the filter coefficients h(n) for M=7

using frequency sampling technique.

5. Design a HPF of length 7 with cut off frequency of 2 rad/sec using Hamming window. Plot the magnitude and phase response. (16)

6. Explain the principle and procedure for designing FIR filter using rectangular window (16)

7. Design a filter with H d (e- j) = e - 3 j , _/4 _ _ _ _/4

0. _/4 _ _ _ _ using a Hanning window with N=7. (16)

8.Design a FIR filter whose frequency response (16)

H (e j ) = 1 _/4 _ _ _ 3 _/4

0. | _ | _3 _/4.Calculate the value of h(n) for N=11 and hence find H(z).

9.Design an ideal differentiator with frequency response H (e j ) = jw -_ _ _ _ _ using hamming window for N=8 and find the frequency response. (16)

10.Design an ideal Hilbert transformer having frequency response

H (e j ) = j -_ _ _ _ 0

-j 0 _ _ _ _ for N=11 using Blackman window. (16)

**FIR structures**:

12.(a) Determine the direct form of following system (8)

$H(z) = 1+2z^{-1} - 3z^{-2} + 4z^{-3} - 5z^{-4}$

(b) Obtain the cascade form realizations of FIR systems (8)

$H(z) = 1+5/2\, z^{-1} + 2z^{-2} + 2\, z^{-3}$

## UNIT - VIII FINITE WORDLENGTH EFFECTS

## PART A ( 2marks)

1. What are the three quantization errors due to finite world length registers in digital filters?(2)

2. What do you mean by limit cycle oscillations? (2)

3. Explain briefly quantization noise. (2)

4. Represent 15.75 in fixed point and in floating point representations. (2)

5.What is the need for scaling in digital filters? (2)

6.List the well known techniques for linear phase FIR filter? (2)

7.What is quantization step size? (2)

8.State the advantages of floating point over fixed point representations? (2)

9.Why rounding preferred over truncation in realizing digital filter? (2)

10.What is meant by dead band? (2).

11.What is over flow limit cycle? How overflow can be eliminated? (2)

12.Sketch the noise probability density functions for rounding? (2)

13.Sketch the noise probability density functions for truncation?. (2)

14.What is meant by finite word length effect in digital filter? (2)

15.Explain the fraction 7/8 and -7/8 in sign magnitude ,1's, 2's complement. (2)

16.Convert in decimal to binary 20.675 (2)

17. Convert in binary to decimal 1110.01 (2)

18. What is product quantization error? (2)

19. What is input quantization error? (2)

20.What is coefficient quantization error? (2)

**PART B**

1. Explain in details about quantization in floating point realizations of IIR filter? (16)

2. Describe the effects of quantization in IIR filter. Consider a first order filter with difference equation y(n) = x(n)+0.5 y(n-1) assume that the data register length is three bits plus a sign bit.

The input x(n) = 0.875 _(n).Explain the limit cycle oscillations in the above filter, if quantization is performed by means of rounding and signed magnitude representation is used.(16)

3. Explain briefly

(1) Effects of coefficient quantization in filter design. (6)

(2) Effects of product round off error in filter design. (6)

(3) Speech recognition (4)

4. Explain briefly

(A)Define limit cycle oscillation. Explain. (8)

(B) Explain the different representation of fixed and floating point representation. (8)

5.Two first order LPF whose system function are given below connected in cascade. Determine

the over all output noise power (16)

$H_1(z) = 1/1-0.9z^{-1}$ and $H_2(z) = 1/1-0.8z^{-1}$

6. (a) Describe the quantization error occur in rounding and truncation in twos complement.(8)

(b) Draw a sample and hold circuit and explains its operation? (8)

7. (a)Explain dead band in limit cycles? (8)

(b)Draw the stastical model of fixed point product quantization and explain (8)

8. (a)What is dead band of a filter? Derive the dead band of second order linear filter? (12)

(b)Consider all pole second order IIR filter described by equation y(n) = -0.5 y(n-1) – 0.75 y(n-2) + x(n).Assuming 8 bits to represent pole, determine the dead band region governing the limit cycle. (4)

9. Determine the variance of the round off noise at the output of two cascaded of the filter with system function $H(z) = H_1(z) . H_2(z)$ where $H_1(z) = 1 / 1-0.5 z^{-1}$ $H_2(z) = 1 / 1-0.25z^{-1}$ (16)

10. Explain with suitable examples the truncation and rounding off errors (16)

11 .a) Explain the application of DSP in Speech processing? (8)

b) What is a vocoder? Explain with a block diagram? (8)

12. Determine the dead band of the filter of y(n) = 0.95 y(n-1) +x(n) (16)

## 20. TUTORIAL PROBLEMS:

1. Find the IDFT of Y (k) = (1, 0, 1, 0)
2. Compute the Fourier transform of the signal x(n) = u(n) – u(n-1).
3. Determine the Discrete Fourier transform x (n) = (1, 1, 1, 1)
4. Derive and draw the 8 point FFT-DIT butterfly structure.
5. Discuss the properties of DFT.
6. Obtain the cascade and parallel form realizations for the following systems (16)
Y (n) = -0.1(n-1) + 0.2 y (n-2) + 3x (n) +3.6 x (n-1) +0.6 x (n-2)
7. Design a HPF of length 7 with cut off frequency of 2 rad/sec using Hamming window. Plot the magnitude and phase response.
8. Convert the analog filter H(s) = 0.5 (s+4) / (s+1)(s+2) using impulse invariant transformation T=0.31416s.

8. Obtain the cascade form realizations of FIR systems

$H(z) = 1+5/2\ z-1+ 2z-2 +2\ z-3$

**21. KNOWN GAPS, IF ANY AND INCLUSION OF THE SAME IN LECTURE SCHEDULE**:

**Known Gaps:**

1. Comparison of received signal with the reference signal (not just by cross correlation but by using time shift parameter).
2. Filters to alter time domain characteristics.
3. To differentiate between the signals based on geographical position of the signal source.
4. To study the systems those are non-linear or not time- invariant.
5. To support much wider range of representing numbers.

**<u>DISCUSSION TOPICS, IF ANY:</u>**

1. Comparae DFS, DFT&DTFT
2. Compare DIT and DIF?
3. Distinguish between DFT and DTFT

4. What do you mean by limit cycle oscillations?
5. Effects of coefficient quantization in filter design.
6. Compare Hamming and Hanning window? (2)

7. What are the advantages of FIR filter?
8. Define canonic and non canonic form realizations

**<u>22.REFERENCESS, JOURNALS, WEBSITES AND E-LINKS IF ANY</u>**:

**<u>References</u>**

AKAIKE, H. 1969. “Power Spectrum Estimation Through Autoregression Model Fitting,” Ann. Inst. Stat. Math., Vol. 21, pp. 407-149

BARTLETT, M. S. 1961. Stochastic Processes, Cambridge University Press,

Cambridge,UK. CLARKE, R. J. 1985. Transform Coding of Images. Academic Press,

London, UK

DYM, H., and McKean , H.P.1972. Fourier Series and Integrals, Academic, New York

**WEBSITES:**

1.www.pearsoned.co.in/johngproakis

2.www.google.com

3.www.wikipedia.com

4.www.analogdevices.com

5.www.dspguru.com



**24. STUDENTS LIST:**

| Class / Section: ECE 31C | | | | | |
|---|---|---|---|---|---|
| **S.No** | **Roll No** | **Student Name** | **S.No** | **Roll No** | **Student Name** |
| 1 | 12R11A04C1 | AKKENAPALLY KIRAN KUMAR | 34 | 12R11A04F4 | MOHAMMAD GOUSEPASHA |
| 2 | 12R11A04C2 | ALLA BALA MURALI KRISHNA | 35 | 12R11A04F5 | MORA SANDEEP REDDY |
| 3 | 12R11A04C3 | ANKIT AGARWAL | 36 | 12R11A04F6 | MORABOINA ASHWINI |
| 4 | 12R11A04C4 | ANNAVARAPU VENKATA SAI KIRAN | 37 | 12R11A04F7 | NAMA VEERA VIGNESHWARA SAI AKHIL KUMAR |
| 5 | 12R11A04C5 | ANUPURAM NARESH | 38 | 12R11A04F8 | PABBOJU DIVYA SREE |
| 6 | 12R11A04C6 | ARUGONDA SAHITHYA | 39 | 12R11A04F9 | P ASHOK RAJU |
| 7 | 12R11A04C7 | BANDI SASIKALA | 40 | 12R11A04G0 | P SAIVARUN REDDY |
| 8 | 12R11A04C8 | B PRAVEEN | 41 | 12R11A04G1 | PAPITHOTTI VIDHYA MADHURI |
| 9 | 12R11A04C9 | BACHALA PRAVEEN KUMAR | 42 | 12R11A04G2 | PASUPULATI JAYASHREE RAO |
| 10 | 12R11A04D0 | DEEPAK REDDY B | 43 | 12R11A04G3 | PINISETTY SRI SAI SRAVANTHI |
| 11 | 12R11A04D1 | BHAVANI TADAM | 44 | 12R11A04G4 | POKKUNURI SAI RAM SUDEEP |
| 12 | 12R11A04D2 | CHERUKUPALLE TANMAI | 45 | 12R11A04G5 | PONUGOTU VISHAL |
| 13 | 12R11A04D3 | DARSHI STEPHEN DEVA RAJ | 46 | 12R11A04G6 | PRIYATHAM REDDY M |
| 14 | 12R11A04D4 | DORNALA VIJAY BHARGAV | 47 | 12R11A04G7 | R KAVYA |
| 15 | 12R11A04D5 | DOSAPATI MOULIKA | 48 | 12R11A04G8 | RAMANI PRIYA KOPALLEY |
| 16 | 12R11A04D6 | G HARIKA | 49 | 12R11A04G9 | SANJAY HARSHITHA SURANA |
| 17 | 12R11A04D7 | GADDAM HARISH | 50 | 12R11A04H0 | SURAPANENI SAI TEJA |
| 18 | 12R11A04D8 | GUGULOTH RAMESH | 51 | 12R11A04H1 | SYED FARHAN |
| 19 | 12R11A04D9 | I UMAMAHESHWARI | 52 | 12R11A04H2 | SYEDA TEHNIYAT |
| 20 | 12R11A04E0 | K KARTHIK | 53 | 12R11A04H3 | T PRAGNYA |

| | | | | | |
|---|---|---|---|---|---|
| 21 | 12R11A04E1 | KALAKONDA ARUN KUMAR | 54 | 12R11A04H4 | TANNERU KALPANA |
| 22 | 12R11A04E2 | KANCHUGANTLA BALARAJU | 55 | 12R11A04H5 | TEJA REDDY CH |
| 23 | 12R11A04E3 | KARKA AMULYA | 56 | 12R11A04H6 | U PAVANI |
| 24 | 12R11A04E4 | KODURU JYOTHIRMAI | 57 | 12R11A04H7 | V N SRIVIDYA |
| 25 | 12R11A04E5 | KOTTE NAVYA | 58 | 12R11A04H8 | V S SRAVAN KUMAR |
| 26 | 12R11A04E6 | LAKHAVATH ANITHA | 59 | 12R11A04H9 | YEDLA RAMESH |
| 27 | 12R11A04E7 | M J AKSHAY KUMAR | 60 | 12R11A04J0 | MUSKU RAJA SANTHOSH REDDY |
| 28 | 12R11A04E8 | M MOUNIKA KRISHNAPRIYA | 61 | 13R15A0411 | AITHA SRIVIDYA |
| 29 | 12R11A04E9 | M S KAMALDEEP | 62 | 13R15A0412 | GARIKIPATI DIVAKAR |
| 30 | 12R11A04F0 | M SUPRAJA | 63 | 13R15A0413 | KURUVA CHANDRASHEKAR |
| 31 | 12R11A04F1 | MADAS SUDHEERA | 64 | 13R15A0414 | NYALAM LAXMAN |
| 32 | 12R11A04F2 | MADHURI M | 65 | 13R15A0415 | KUKKADHUVULA PANDU |
| 33 | 12R11A04F3 | MEKALA ISHWARYA | | | |
| | | | | | |
| Total: 65 Males: 38 Females: 27 | | | | | |

## 25.QUALITY MEASUREMENT SHEETS

1. COURSE END SUREY
2. TEACHING EVALUATION

## 26.GROUP- WISE STUDENTS LIST FOR DISCUSSION TOPICS:

## GROUP 1

| | | |
|---|---|---|
| 1 | 12R11A04C1 | AKKENAPALLY KIRAN KUMAR |
| 2 | 12R11A04C2 | ALLA BALA MURALI KRISHNA |
| 3 | 12R11A04C3 | ANKIT AGARWAL |
| 4 | 12R11A04C4 | ANNAVARAPU VENKATA SAI KIRAN |
| 5 | 12R11A04C5 | ANUPURAM NARESH |
| 6 | 12R11A04C6 | ARUGONDA SAHITHYA |

## GROUP 2

| | | |
|---|---|---|
| 7 | 12R11A04C7 | BANDI SASIKALA |
| 8 | 12R11A04C8 | B PRAVEEN |
| 9 | 12R11A04C9 | BACHALA PRAVEEN KUMAR |
| 10 | 12R11A04D0 | DEEPAK REDDY B |
| 11 | 12R11A04D1 | BHAVANI TADAM |
| 12 | 12R11A04D2 | CHERUKUPALLE TANMAI |
| 13 | 12R11A04D3 | DARSHI STEPHEN DEVA RAJ |

## GROUP 3

| | | |
|---|---|---|
| 13 | 12R11A04D3 | DARSHI STEPHEN DEVA RAJ |

## GROUP 7

| | | |
|---|---|---|
| 34 | 12R11A04F4 | MOHAMMAD GOUSEPASHA |
| 35 | 12R11A04F5 | MORA SANDEEP REDDY |
| 36 | 12R11A04F6 | MORABOINA ASHWINI |
| 37 | 12R11A04F7 | NAMA VEERA VIGNESHWARA SAI AKHIL KUMAR |
| 38 | 12R11A04F8 | PABBOJU DIVYA SREE |
| 39 | 12R11A04F9 | P ASHOK RAJU |

## GROUP 8

| | | |
|---|---|---|
| 40 | 12R11A04G0 | P SAIVARUN REDDY |
| 41 | 12R11A04G1 | PAPITHOTTI VIDHYA MADHURI |
| 42 | 12R11A04G2 | PASUPULATI JAYASHREE RAO |
| 43 | 12R11A04G3 | PINISETTY SRI SAI SRAVANTHI |
| 44 | 12R11A04G4 | POKKUNURI SAI RAM SUDEEP |
| 45 | 12R11A04G5 | PONUGOTU VISHAL |
| 46 | 12R11A04G6 | PRIYATHAM REDDY M |

## GROUP 9

| | | |
|---|---|---|
| 46 | 12R11A04G6 | PRIYATHAM REDDY M |

| | | |
|---|---|---|
| 14 | 12R11A04D4 | DORNALA VIJAY BHARGAV |
| 15 | 12R11A04D5 | DOSAPATI MOULIKA |
| 16 | 12R11A04D6 | G HARIKA |
| 17 | 12R11A04D7 | GADDAM HARISH |
| 18 | 12R11A04D8 | GUGULOTH RAMESH |

## GROUP 4

| | | |
|---|---|---|
| 19 | 12R11A04D9 | I UMAMAHESHWARI |
| 20 | 12R11A04E0 | K KARTHIK |
| 21 | 12R11A04E1 | KALAKONDA ARUN KUMAR |
| 22 | 12R11A04E2 | KANCHUGANTLA BALARAJU |
| 23 | 12R11A04E3 | KARKA AMULYA |
| 24 | 12R11A04E4 | KODURU JYOTHIRMAI |

## GROUP5

| | | |
|---|---|---|
| 25 | 12R11A04E5 | KOTTE NAVYA |
| 26 | 12R11A04E6 | LAKHAVATH ANITHA |
| 27 | 12R11A04E7 | M J AKSHAY KUMAR |
| 28 | 12R11A04E8 | M MOUNIKA KRISHNAPRIYA |
| 29 | 12R11A04E9 | M S KAMALDEEP |
| 30 | 12R11A04F0 | M SUPRAJA |

## GROUP6

| | | |
|---|---|---|
| 47 | 12R11A04G7 | R KAVYA |
| 48 | 12R11A04G8 | RAMANI PRIYA KOPALLEY |
| 49 | 12R11A04G9 | SANJAY HARSHITHA SURANA |
| 50 | 12R11A04H0 | SURAPANENI SAI TEJA |
| 51 | 12R11A04H1 | SYED FARHAN |

## GROUP 10

| | | |
|---|---|---|
| 52 | 12R11A04H2 | SYEDA TEHNIYAT |
| 53 | 12R11A04H3 | T PRAGNYA |
| 54 | 12R11A04H4 | TANNERU KALPANA |
| 55 | 12R11A04H5 | TEJA REDDY CH |
| 56 | 12R11A04H6 | U PAVANI |
| 57 | 12R11A04H7 | V N SRIVIDYA |

## GROUP11

| | | |
|---|---|---|
| 58 | 12R11A04H8 | V S SRAVAN KUMAR |
| 59 | 12R11A04H9 | YEDLA RAMESH |
| 60 | 12R11A04J0 | MUSKU RAJA SANTHOSH REDDY |
| 61 | 13R15A0411 | AITHA SRIVIDYA |
| 62 | 13R15A0412 | GARIKIPATI DIVAKAR |
| 63 | 13R15A0413 | KURUVA CHANDRASHEKAR |

| | | | | | |
|---|---|---|---|---|---|
| 31 | 12R11A04F1 | MADAS SUDHEERA | 64 | 13R15A0414 | NYALAM LAXMAN |
| 32 | 12R11A04F2 | MADHURI M | 65 | 13R15A0415 | KUKKADHUVULA PANDU |
| 33 | 12R11A04F3 | MEKALA ISHWARYA | | | |