

# APPLIED DIGITAL SIGNAL PROCESSING AND APPLICATIONS



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## Preface

Due to the rapid development of technologies, digital information playing a key role in our daily life. In the past signal processing appeared in various concepts in more traditional courses where the analog and discrete components were used to achieve the various objectives. However, in the 21<sup>st</sup> century, with the rapid growth of computing power in terms of speed and memory capacity and the intervention of artificial intelligent, machine /deep learning algorithms introduces a tremendous growth in signal processing applications. Therefore, digital signal processing has become such a critical component in contemporary science and technology that many tasks would not be attempted without it. It is a truly interdisciplinary subject that draws from synergistic developments involving many disciplines. The developers should be able to solve problems with an innovation, creativity and active initiators of novel ideas. However, the learning and teaching has been changed from conventional and tradition education to outcome based education. Therefore, this book prepared on a Problem-based approach and outcome based education strategies. Where the problems incorporate most of the basic principles and proceeds towards implementation of more complex algorithms. Students required to formulate in a way to achieve a well-defined goals under the guidance of their instructor.

This book follows a holistic approach and presents discrete-time processing as a seamless continuation of continuous-time signals and systems, beginning with a review of continuous-time signals and systems, frequency response, and filtering. The synergistic combination of continuous-time and discrete-time perspectives leads to a deeper appreciation and understanding of DSP concepts and practices.

This book is organized in Ten chapters as follows: Chapter One, introduces the basic terminology of signals in digital signal processing. Classification of signals as well as the elementary signal are explained in detail. Chapter Two describes the concept of systems and characterize and analyze the properties of Discrete systems. Chapter Three covers the sampling process, Quantization, coding and reconstruction of signals. Chapter Four introduces the properties of discrete signals and systems. Chapter Five introduces the z-transform and difference equations and its applications. Chapter Six explains the frequency analysis of Discrete Signals and Systems, Frequency Response of Systems and convolution via frequency domain. Chapter Seven devoted for Discrete Fourier transform. Chapter Eight deals with various methods used in Digital filters design. Chapter Nine introduces the wavelet transforms, Multiresolution Analysis and some applications of discrete wavelet transform. Chapter Ten deals with adaptive signal processing and covers Wiener filter, LMS algorithms, RLS algorithms and ends with applications of adaptive filters.

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# Contents

[Preface](#)

[Dedication](#)

## [Chapter 1 Introduction to Signals](#)

[1.1 Introduction](#)

[1.2 Signal Classification](#)

[1.2.1 Continuity of the independent and dependent variables](#)

[1.2.2 Predictability of the dependent variables with respect to the independent variable.](#)

[1.2.3 Dimensionality of Signals](#)

[1.2.4 Periodic vs. Aperiodic Signals](#)

[1.2.5 Causal vs. Anticausal Signals](#)

[1.2.6 Even vs. Odd Signals](#)

[1.2.7 Energy vs. Power Signals](#)

[1.3 Elementary Signals](#)

[1.3.1 Unit Impulse Function](#)

[1.3.2 Unit Step Function](#)

[1.3.3 Rectangular Pulse Function](#)

[1.3.4 Signum function](#)

[1.3.5 Ramp function](#)

[1.3.6 Sinc function](#)

[1.3.7 Exponential Function](#)

## [Chapter 2 Introduction to Systems](#)

[2.1 Introduction](#)

[2.2 Classification of Systems](#)

[2.2.1 Linear and non-linear systems](#)

[2.2.2 Time-varying and time-invariant systems](#)

[2.2.3 Static and Dynamic Systems](#)

[2.2.4 Invertible and non-invertible systems](#)

[2.2.5 Causal and non-causal systems](#)

[2.2.6 Stable and unstable systems](#)

[2.3 Impulse Response and Convolution](#)

## [Chapter 3 Sampling, Quantization and Reconstruction](#)

[3.1 Introduction](#)

[3.2 Signal Sampling](#)

[3.3 Interpolation](#)

[3.4 The Sampling Theorem](#)

[3.5 Aliasing](#)

[3.6 Antialiasing Prefilters](#)

[3.7 Types of Sampling](#)

[3.7.1 Impulse \(Ideal\) Sampling](#)

[3.7.2 Natural Sampling](#)

[3.7.3 Sample-and-Hold \(Flat Top\) Sampling](#)

[3.8 Quantization](#)

[3.8.1 Quantization Error](#)

[3.9 Ideal Reconstruction](#)

[3.10 Signal Reconstruction](#)

## Chapter 4 Discrete-Time Signals and Systems

### 4.1 Discrete-Time Signals

#### 4.1.1 Some Elementary Sequences

##### 4.1.1.1 Unit Impulse Sequence

##### 4.1.1.2 Unit Step Sequence

##### 4.1.1.3 The unit ramp signal

##### 4.1.1.4 Sinusoidal Sequences

##### 4.1.1.5 Complex Exponential Sequences

##### 4.1.1.6 Random Sequences

#### 4.2 Types of Sequences

##### 4.2.1 Real vs. Complex Signals

##### 4.2.2 Finite vs. Infinite Length

##### 4.2.3 Causal vs. Anti-causal Signals

##### 4.2.4 Energy and Power Signals

#### 4.3 Some Basic Operations on Sequences

### 4.4 Discrete-time Systems

#### 4.4.1 Classification of Systems

#### 4.4.2 Linear Shift-Invariant Systems

#### 4.4.3 Linear Convolution

#### 4.4.4 Properties of Linear Convolution

##### 4.4.4.1 Condition for Stability

##### 4.4.4.2 Condition for Causality

## Chapter 5 Z-transform and applications

### 5.1 Introduction

### 5.2 Unilateral Z-transform

### 5.3 Bilateral Z-transform

### 5.4 Poles and Zeros in the Z-Plane

### 5.5 Properties of the z transform

### 5.6 Region of Convergence for the Z-Transform

#### 5.6.1 Properties of the Region of Convergence

### 5.7 Inverse z-Transform

#### 5.7.1 Power Series Method

#### 5.7.2 Partial Fraction Expansion

#### 5.7.3 Contour integration.

### 5.8 Transfer Function in the Z-domain

### 5.9 Application to signal processing

#### 5.9.1 Solution of Difference Equations Using the z-Transform

#### 5.9.2 Analysis of Linear Discrete Systems

## Chapter 6 Frequency Analysis of Discrete Signals and Systems

### 6.1 Introduction

### 6.2 Frequency analysis of a Continuous Time signal

#### 6.2.1 Fourier Series for Continuous-Time Periodic Signals

### 6.3 Frequency Analysis of Discrete-Time Signals

#### 6.3.1 Fourier Series for Discrete-Time Periodic Signals

#### 6.3.2 Fourier Transform of Discrete-Time Aperiodic Signals

### 6.4 Frequency Domain Representation of Discrete-time LTI Systems

#### 6.4.1 Steady State Response of LTI Discrete-time Systems

### 6.5 Frequency Response of Systems

### 6.6 Convolution via the Frequency Domain

## Chapter 7 Discrete Fourier Transform

### 7.1 Introduction

## 7.2 DFT as matrix multiplication

## 7.3 Properties of the DFT

### 7.3.1 Periodicity

### 7.3.2 Orthogonality

### 7.3.3 Linearity

### 7.3.4 Hermitian symmetry

### 7.3.5 Time shifting

### 7.3.6 Circular convolution

### 7.3.7 Parseval's theorem

## 7.4 Computational complexity

## 7.5 Fast Fourier Transform (FFT)

### 7.5.1 Derivation of the FFT

## Chapter 8 Design of Digital Filters

### 8.1 Introduction

#### 8.1.1. Finite Impulse Response

#### 8.1.2 Infinite Impulse Response

#### 8.1.3 Filter Specification Requirements

### 8.2 FIR Digital Filters

#### 8.2.1 Design of FIR Digital Filters using Impulse Response Truncation (IRT)

#### 8.2.2 Design of FIR filters using windowing technique.

#### 8.2.3 Design of FIR filters by frequency sampling

### 8.3 Design of IIR Filters

#### 8.3.1 IIR Filter Basics

#### 8.3.2 Bilinear transformation method

#### 8.3.3 Analog Filter using lowpass prototype Transformation

#### 8.3.4 Bilinear Transformation and Frequency Warping

#### 8.3.5 Bilinear Transformation Design Procedure

#### 8.4.6 Impulse Invariant Design Method

## Chapter 9 Wavelet Transform

### 9.1 Introduction

### 9.2 Continuous Wavelet Transform

### 9.3 Time-Frequency Resolution

### 9.4 Wavelet Series

#### 9.4.1 Dyadic Sampling

### 9.5 Discrete Wavelet Transform (DWT)

#### 9.5.1 Multiresolution Analysis

#### 8.5.2 Wavelet Analysis by Multirate Filtering

#### 8.5.3 Wavelet Synthesis by Multirate Filtering

### 9.6 Discrete Wavelet Transform for denoising data

### 9.7 Signal denoising for IoT networks

### 9.8 Multiresolution Signal Analysis

### 9.9 Multiresolution Wavelet Decomposition of Transient Signal

### 9.10 Signal Detection

## Chapter 10 Adaptive Signal Processing

### 10.1 Introduction

### 10.2 Adaptive Noise Cancellation

### 10.3 Adaptive Filtering Algorithms

#### 10.3.1 Least Mean Square (LMS) Algorithm

#### 10.3.2 The Recursive Least Squares (RLS) Algorithm

#### 10.3.3 Wiener Filtering

##### 10.3.3.1 Adaptive Wiener Filter

## 10.4 Applications of Adaptive Filters

### 10.4.1 System Identification

### 10.4.2 Channel Identification

### 10.4.3 Plant Identification

### 10.4.4 Echo Cancellation for Long-Distance Transmission

### 10.4.5 Acoustic Echo Cancellation

### 10.4.6 Adaptive Noise Cancelling

## 10.5 Inverse Modeling

### 10.5.1 Channel Equalization

### 10.5.2 Inverse Plant Modeling

### 10.5.3 Linear Prediction

#### 10.5.3.1 Linear Predictive Coding

### 10.5.4 Adaptive Line Enhancement

## 10.6 Adaptive Noise Reduction

## References