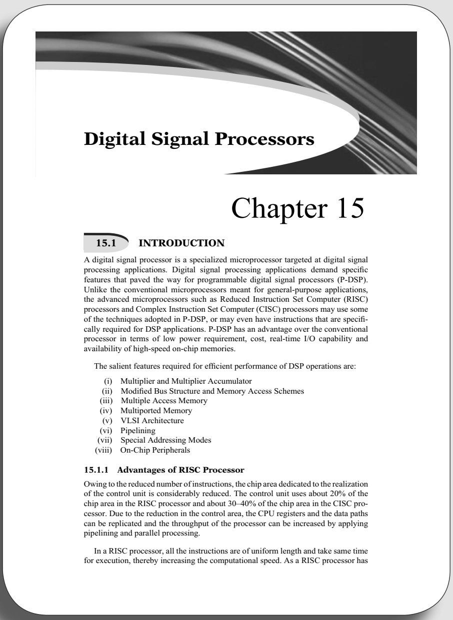
VISUAL WALKTHROUGH

New Chapter

A new chapter on Digital Signal Processors is introduced in this edition.



Digital Signal Processors

Chapter 15

15.1 INTRODUCTION

A digital signal processor is a specialized microprocessor targeted at digital signal processing applications. Digital signal processing applications demand specific features that paved the way for programmable digital signal processors (P-DSP). Unlike the conventional microprocessors meant for general-purpose applications, the advanced microprocessors such as Reduced Instruction Set Computer (RISC) processors and Complex Instruction Set Computer (CISC) processors may use some of the techniques adopted in P-DSP, or may even have instructions that are specifically required for DSP applications. P-DSP has an advantage over the conventional processor in terms of low power requirement, cost, real-time I/O capability and availability of high-speed on-chip memories.

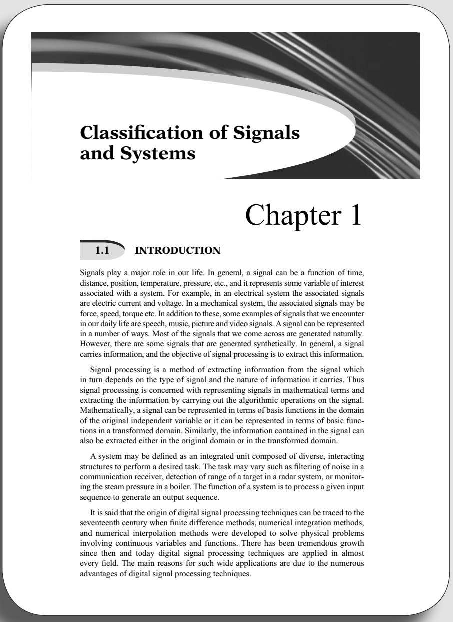
The salient features required for efficient performance of DSP operations are:

(i) Multiplier and Multiplier Accumulator
(ii) Modified Bus Structure and Memory Access Schemes
(iii) Multiple Access Memory
(iv) Multiported Memory
(v) VLSI Architecture
(vi) Pipelining
(vii) Special Addressing Modes
(viii) On-Chip Peripherals

15.1.1 Advantages of RISC Processor

Owing to the reduced number of instructions, the chip area dedicated to the realization of the control unit is considerably reduced. The control unit uses about 20% of the chip area in the RISC processor and about 30–40% of the chip area in the CISC processor. Due to the reduction in the control area, the CPU registers and the data paths can be replicated and the throughput of the processor can be increased by applying pipelining and parallel processing.

In a RISC processor, all the instructions are of uniform length and take same time for execution, thereby increasing the computational speed. As a RISC processor has



Classification of Signals and Systems

Chapter 1

1.1 INTRODUCTION

Signals play a major role in our life. In general, a signal can be a function of time, distance, position, temperature, pressure, etc., and it represents some variable of interest associated with a system. For example, in an electrical system the associated signals are electric current and voltage. In a mechanical system, the associated signals may be force, speed, torque etc. In addition to these, some examples of signals that we encounter in our daily life are speech, music, picture and video signals. A signal can be represented in a number of ways. Most of the signals that we come across are generated naturally. However, there are some signals that are generated synthetically. In general, a signal carries information, and the objective of signal processing is to extract this information.

Signal processing is a method of extracting information from the signal which in turn depends on the type of signal and the nature of information it carries. Thus signal processing is concerned with representing signals in mathematical terms and extracting the information by carrying out the algorithmic operations on the signal. Mathematically, a signal can be represented in terms of basis functions in the domain of the original independent variable or it can be represented in terms of basic functions in a transformed domain. Similarly, the information contained in the signal can also be extracted either in the original domain or in the transformed domain.

A system may be defined as an integrated unit composed of diverse, interacting structures to perform a desired task. The task may vary such as filtering of noise in a communication receiver, detection of range of a target in a radar system, or monitoring the steam pressure in a boiler. The function of a system is to process a given input sequence to generate an output sequence.

It is said that the origin of digital signal processing techniques can be traced to the seventeenth century when finite difference methods, numerical integration methods, and numerical interpolation methods were developed to solve physical problems involving continuous variables and functions. There has been tremendous growth since then and today digital signal processing techniques are applied in almost every field. The main reasons for such wide applications are due to the numerous advantages of digital signal processing techniques.

Introduction

Each chapter begins with an Introduction that gives a brief summary of the background and contents of the chapter.

Advantages of representing the digital system in block diagram form

(i) Just by inspection, the computation algorithm can be easily written
(ii) The hardware requirements can be easily determined
(iii) A variety of equivalent block diagram representations can be easily developed from the transfer function
(iv) The relationship between the output and the input can be determined.

9.2.1 Canonic and Non-Canonic Structures

If the number of delays in the realisation block diagram is equal to the order of the difference equation or the order of the transfer function of a digital filter, then the realisation structure is called **canonic**. Otherwise, it is a **non-canonic** structure.

9.3 BASIC STRUCTURES FOR IIR SYSTEMS

Causal IIR systems are characterised by the constant coefficient difference equation of Eq. 9.1 or equivalently, by the real rational transfer function of Eq. 9.2. From these equations, it can be seen that the realisation of infinite duration impulse response (IIR) systems involves a recursive computational algorithm. In this section, the most important filter structures namely direct Forms I and II, cascade and parallel realisations for IIR systems are discussed.

9.3.1 Direct Form Realisation of IIR Systems

Equation 9.2 is the standard form of the system transfer function. By inspection of this equation, the block diagram representation can be drawn directly for the **direct form realisation**. The multipliers in the feed forward paths are the numerator coefficients and the multipliers in the feedback paths are the negatives of the denominator coefficients. Since the multiplier coefficients in the structures are exactly the coefficients of the transfer function, they are called direct form structures.

Direct Form I

The digital system structure determined directly from either Eq. 9.1 or Eq. 9.2 is called the direct form I. In this case, the system function is divided into two parts connected in cascade, the first part containing only the zeros, followed by the part containing only the poles. An intermediate sequence $w(n)$ is introduced. A possible IIR system direct form I realisation is shown in Fig. 9.3, in which $w(n)$ represents the output of the first part and input to the second.

Sections and Sub-sections

Each chapter has been neatly divided into sections and sub-sections so that the subject matter is studied in a logical progression of ideas and concepts.

Finite Impulse Response (FIR) Filters

Chapter 7

Infinite Impulse Response (IIR) Filters

Chapter 8

Effects of Finite Word Length in Digital Filters

Chapter 10

Multirate Digital Signal Processing

Chapter 11

Applications of Digital Signal Processing

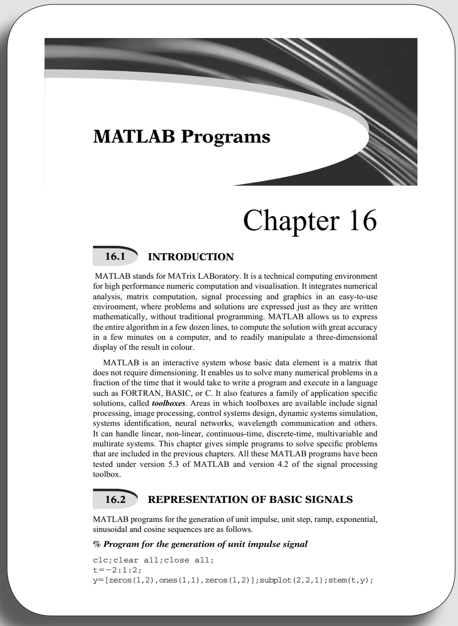
Chapter 14

Coverage of Topics

- In depth coverage of key topics including *IIR Filters, FIR Filters, Effect of Finite Word Length Effects in Digital Filters* and *Multirate Digital Signal Processing*
- Review of essential mathematical concepts like *Fourier Transforms, Laplace Transforms* and *z Transforms.*
- Detailed treatment on Applications of Digital Signal Processing

MATLAB

A separate chapter for MATLAB is included containing 45 MATLAB examples



MATLAB Programs

Chapter 16

16.1 INTRODUCTION

MATLAB stands for MATrix LABoratory. It is a technical computing environment for high performance numeric computation and visualisation. It integrates numerical analysis, matrix computation, signal processing and graphics in an easy-to-use environment, where problems and solutions are expressed just as they are written mathematically, without traditional programming. MATLAB allows us to express the entire algorithm in a few dozen lines, to compute the solution with great accuracy in a few minutes on a computer, and to readily manipulate a three-dimensional display of the result in colour.

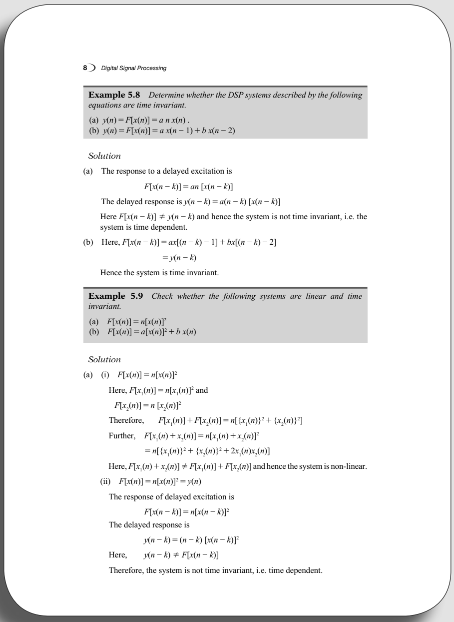
MATLAB is an interactive system whose basic data element is a matrix that does not require dimensioning. It enables us to solve many numerical problems in a fraction of the time that it would take to write a program and execute in a language such as FORTRAN, BASIC, or C. It also features a family of application specific solutions, called ***toolboxes***. Areas in which toolboxes are available include signal processing, image processing, control systems design, dynamic systems simulation, systems identification, neural networks, wavelength communication and others. It can handle linear, non-linear, continuous-time, discrete-time, multivariable and multirate systems. This chapter gives simple programs to solve specific problems that are included in the previous chapters. All these MATLAB programs have been tested under version 5.3 of MATLAB and version 4.2 of the signal processing toolbox.

16.2 REPRESENTATION OF BASIC SIGNALS

MATLAB programs for the generation of unit impulse, unit step, ramp, exponential, sinusoidal and cosine sequences are as follows.

% Program for the generation of unit impulse signal

```
clc;clear all;close all;
t=−2:1:2;
y=[zeros(1,2),ones(1,1),zeros(1,2)];subplot(2,2,1);stem(t,y);
```



Example 5.8 *Determine whether the DSP systems described by the following equations are time invariant.*

(a) $y(n) = F[x(n)] = a\,n\,x(n)$.
(b) $y(n) = F[x(n)] = a\,x(n-1) + b\,x(n-2)$

Solution

(a) The response to a delayed excitation is

$$F[x(n-k)] = an\,[x(n-k)]$$

The delayed response is $y(n-k) = a(n-k)\,[x(n-k)]$

Here $F[x(n-k)] \neq y(n-k)$ and hence the system is not time invariant, i.e. the system is time dependent.

(b) Here, $F[x(n-k)] = ax[(n-k)-1] + bx[(n-k)-2]$

$$= y(n-k)$$

Hence the system is time invariant.

Example 5.9 *Check whether the following systems are linear and time invariant.*

(a) $F[x(n)] = n[x(n)]^2$
(b) $F[x(n)] = a[x(n)]^2 + b\,x(n)$

Solution

(a) (i) $F[x(n)] = n[x(n)]^2$

Here, $F[x_1(n)] = n[x_1(n)]^2$ and

$F[x_2(n)] = n\,[x_2(n)]^2$

Therefore, $F[x_1(n)] + F[x_2(n)] = n[\{x_1(n)\}^2 + \{x_2(n)\}^2]$

Further, $F[x_1(n) + x_2(n)] = n[x_1(n) + x_2(n)]^2$

$= n[\{x_1(n)\}^2 + \{x_2(n)\}^2 + 2x_1(n)x_2(n)]$

Here, $F[x_1(n) + x_2(n)] \neq F[x_1(n)] + F[x_2(n)]$ and hence the system is non-linear.

(ii) $F[x(n)] = n[x(n)]^2 = y(n)$

The response of delayed excitation is

$$F[x(n-k)] = n[x(n-k)]^2$$

The delayed response is

$$y(n-k) = (n-k)\,[x(n-k)]^2$$

Here, $y(n-k) \neq F[x(n-k)]$

Therefore, the system is not time invariant, i.e. time dependent.

Worked Examples

Numerous Worked Examples, totaling to over 250, are provided in sufficient number in each chapter and at appropriate locations, to aid in understanding of the text material.

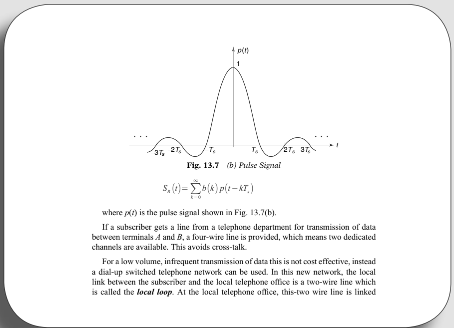


Fig. 13.7 *(b) Pulse Signal*

$$S_B(t) = \sum_{k=0}^{\infty} b(k)\, p(t - kT_s)$$

where $p(t)$ is the pulse signal shown in Fig. 13.7(b).

If a subscriber gets a line from a telephone department for transmission of data between terminals A and B, a four-wire line is provided, which means two dedicated channels are available. This avoids cross-talk.

For a low volume, infrequent transmission of data this is not cost effective, instead a dial-up switched telephone network can be used. In this new network, the local link between the subscriber and the local telephone office is a two-wire line which is called the ***local loop***. At the local telephone office, this-two wire line is linked

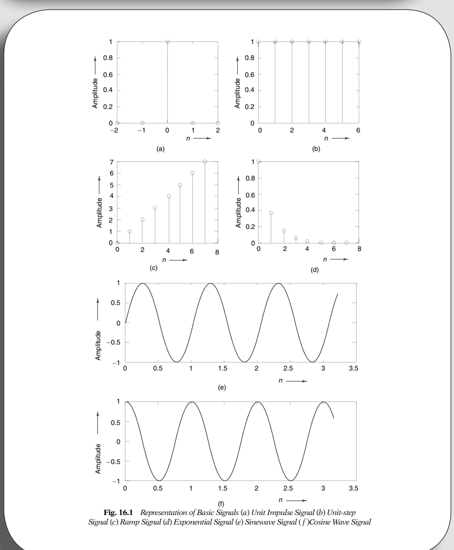


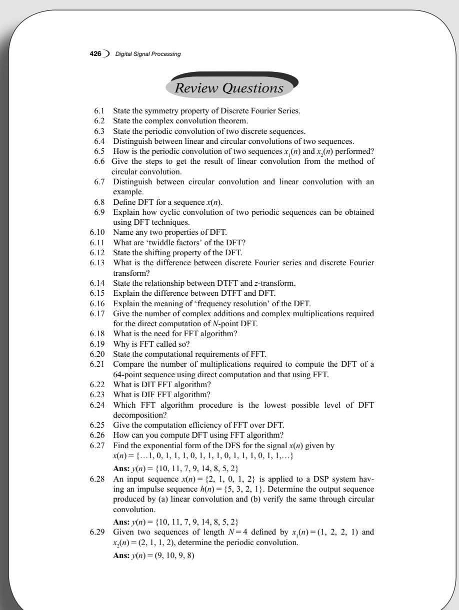
Fig. 16.1 *Representation of Basic Signals (a) Unit Impulse Signal (b) Unit-step Signal (c) Ramp Signal (d) Exponential Signal (e) Sinewave Signal (f) Cosine Wave Signal*

Illustrations

Illustrations are essential tools in books on Engineering subjects. Ample illustrations are provided in each chapter to illustrate the concepts, functional relationships and to provide definition sketches for mathematical models.

Review Questions

Each chapter contains a set of Review Questions, totaling to over 600 problems in the book. Solution to these requires not only application of the material covered in the book but also enables the student to strive towards good comprehension of the subject matter.



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Review Questions

6.1 State the symmetry property of Discrete Fourier Series.
6.2 State the complex convolution theorem.
6.3 State the periodic convolution of two discrete sequences.
6.4 Distinguish between linear and circular convolutions of two sequences.
6.5 How is the periodic convolution of two sequences $x_1(n)$ and $x_2(n)$ performed?
6.6 Give the steps to get the result of linear convolution from the method of circular convolution.
6.7 Distinguish between circular convolution and linear convolution with an example.
6.8 Define DFT for a sequence $x(n)$.
6.9 Explain how cyclic convolution of two periodic sequences can be obtained using DFT techniques.
6.10 Name any two properties of DFT.
6.11 What are 'twiddle factors' of the DFT?
6.12 State the shifting property of the DFT.
6.13 What is the difference between discrete Fourier series and discrete Fourier transform?
6.14 State the relationship between DTFT and z-transform.
6.15 Explain the difference between DTFT and DFT.
6.16 Explain the meaning of 'frequency resolution' of the DFT.
6.17 Give the number of complex additions and complex multiplications required for the direct computation of N-point DFT.
6.18 What is the need for FFT algorithm?
6.19 Why is FFT called so?
6.20 State the computational requirements of FFT.
6.21 Compare the number of multiplications required to compute the DFT of a 64-point sequence using direct computation and that using FFT.
6.22 What is DIT FFT algorithm?
6.23 What is DIF FFT algorithm?
6.24 Which FFT algorithm procedure is the lowest possible level of DFT decomposition?
6.25 Give the computation efficiency of FFT over DFT.
6.26 How can you compute DFT using FFT algorithm?
6.27 Find the exponential form of the DFS for the signal $x(n)$ given by $x(n) = \{\ldots 1, 0, 1, 1, 1, 0, 1, 1, 1, 0, 1, 1, 1, 0, 1, 1, \ldots\}$

Ans: $y(n) = \{10, 11, 7, 9, 14, 8, 5, 2\}$

6.28 An input sequence $x(n) = \{2, 1, 0, 1, 2\}$ is applied to a DSP system having an impulse sequence $h(n) = \{5, 3, 2, 1\}$. Determine the output sequence produced by (a) linear convolution and (b) verify the same through circular convolution.

Ans: $y(n) = \{10, 11, 7, 9, 14, 8, 5, 2\}$

6.29 Given two sequences of length $N = 4$ defined by $x_1(n) = (1, 2, 2, 1)$ and $x_2(n) = (2, 1, 1, 2)$, determine the periodic convolution.

Ans: $y(n) = (9, 10, 9, 8)$