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Digital Signal Processing

(Third Edition)

数字信号处理

（第三版）

〔印〕 S Poornachandra
B Sasikala 著

（英文影印版）

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内 容 简 介

本书主要面向数字信号处理的初学者。内容主要涵盖了离散傅里叶变换、数字滤波器、有限字长和多频信号处理等。全书风格明晰，提供了很多问题解决方案、插图及流程图，非常有助于读者入门。

本书可作为信息类专业本科生学习数字信号处理的双语教材或参考书，也可供工程技术人员参考。

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Preface

This book *Digital Signal Processing*, *3e* is designed for a one-semester course on the subject for students of engineering at the undergraduate level. The book has its emphasis on ease of comprehension, lucid explanation of concepts, numerous examples, a solved problems approach and simple presentation.

The first four chapters of the book provides an introduction and recap of the topics on signals and systems. Students who have studied Signals and Systems as a core paper may decide to skip these chapters.

A significant highlight of this book is the treatment and coverage of topics on Finite Impulse Response Filters (FIR) and Infinite Impulse Response Filter (IIR). It reckons for students who find the subject tough and provides numerous examples with explanations.

Similarly, the topic Finite World Length Effect has its emphasis on clear concepts and a simple and easy to understand presentation. The coverage of the topic has a prerequisite that the students are familiar with number and decimal systems.

The topic Multirate Signal Processing has been discussed with necessary mathematical treatment. The basic concepts are explained in simple English to facilitate better comprehension.

This textbook has a crisp and clear introduction to Estimation Theory starting from Estimation Parameters to Model Estimation.

The field of Digital Signal Processing has its impact on all areas of technology and science. It is of equal importance to industry and academia. In the engineering curriculum, this subject is now offered to students of electronics, electrical, communication, IT and computer science streams. We hope this book will serve as a basic resource for all students of engineering.

We thank the management and staff members of our respective institutions for all their support and help. We thank Mr. P K Madhavan and others of Vijay Nicole for their efficient and tireless efforts in publishing this book.

We welcome all objective criticisms and suggestions on the book.

Dr. S Poornachandra
B Sasikala

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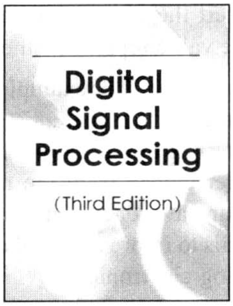


CHAPTER1

Introduction to Digital Signal Processing

DSP or Digital Signal Processing, as the term suggests, is the processing of signals by digital means. A signal in this context means a source of information. In general terms, a signal is a stream of information representing anything from stock prices to data from a remote-sensing satellite. The signal here means an electrical signal carried by a wire or telephone line, or perhaps by a radio wave. In many cases, the signal is initially in the form of an analog electrical voltage or current, produced for example by a microphone or some other type of transducer. In some situations the data is already in digital form—such as the output from the readout system of a CD (compact disc) player. An analog signal must be converted into digital (i.e. numerical) form before DSP techniques can be applied. An analog electrical voltage signal, for example, can be digitized using an analog-to-digital converter (ADC). An analog signal on sampling results in a discrete signal followed by quantization and encoding in order to convert the discrete signal to digital signal. This generates a digital output in the form of a binary number whose value represents the electrical voltage input to the device.

Signals need to be processed in a variety of ways. For example, the output signal from a transducer may be contaminated with noise. The electrodes attached to a patient's chest when an electrocardiogram (ECG) is taken, measure tiny electrical voltage changes due to the activity of the heart and other muscles. The signal is often strongly affected due to electrical interference from the mains supply, electromagnetic interference, muscle artifacts, etc. Processing the signal using a filter circuit can remove or at least reduce the unwanted part of the signal. Nowadays, the filtering of signals to improve signal quality or to extract important information is done by DSP techniques rather than by analog electronics.

The development of digital signal processing dates from the 1960s with the use of mainframe digital computers for number-crunching applications such as the Fast Fourier Transform (FFT), which allows the frequency spectrum of a signal to be computed rapidly. These techniques were not widely used earlier because suitable computing equipment was available only in leading universities and other scientific research institutions. The introduction of the microprocessor in the late 1970s and early 1980s made it possible for DSP techniques to be used in a much wider range of applications. However, general-purpose microprocessors such as the Intel x86 family are not ideally suited to the numerically-intensive requirements of DSP, and during the 1980s the increasing importance of DSP led several major electronic manufacturers (such as Texas Instruments, Analog Devices, and Motorola) to develop Digital Signal Processor chips—specialized microprocessors with architectures designed specifically for the types of operations required in digital signal processing. (Note that the acronym DSP can variously mean Digital Signal Processing, the term used for a wide range of techniques for processing signals digitally, or Digital Signal Processor, a specialized type of microprocessor chips). Like a general-purpose microprocessor, a DSP is a programmable device, with its own native instruction code. DSP chips are capable of carrying out millions of floating point operations per second, and like their better-known general-purpose cousins, faster and more powerful versions are continually being introduced.

DSP technology is commonly employed nowadays in devices such as mobile phones, multimedia computers, video recorders, CD players, hard disc drive controllers and modems, and will soon replace analog circuitry in TV sets and telephones. An important application of DSP is in signal compression and decompression. In CD systems, for example, the music recorded on the CD is in a compressed form (to increase storage capacity) and must be decompressed for the recorded signal to be reproduced. Signal compression is used in digital cellular phones to allow a greater number of calls to be handled simultaneously within each local "cell". DSP signal compression technology allows people not only to talk to one another by telephone but also to see one another on the screens of their PCs, using small video cameras mounted on the computer monitors, with only a conventional telephone line linking them together. Although the mathematical theory underlying DSP techniques such as Fast Fourier transform, Wavelet transform, Hilbert transform, Digital filter design and Signal compression can be fairly complex, the numerical operations required to implement these techniques are in fact very simple, consisting mainly of operations that could be done on a cheap four-function calculator. The architecture of a DSP chip is designed to carry out such operations incredibly fast, processing upto tens of millions of samples per second, to provide real-time performance, that is, the ability to process a signal "live" as it is sampled and then output the processed signal, for example, to a loud speaker or video display. All the practical applications DSP mentioned earlier, such as hard disc drives and mobile phones, demand real-time operation.

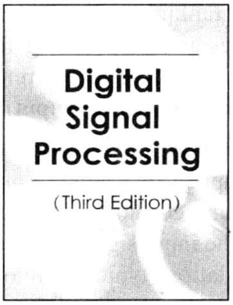
In signal processing, the function of a filter is to remove unwanted parts of the signal, such as random noise, or to extract useful parts of the signal, such as the components lying within a certain frequency range. There are two main kinds of filters, analog and digital. They are quite different in their physical makeup and in their working. An analog filter uses analog electronic circuits made from components such as resistors, capacitors and op amps to produce the required filtering effect. Such filter circuits are widely used in applications such as noise reduction, video signal enhancement, graphic equalizers in *hi-fi* systems, and many other areas. There are well-established standard techniques for designing an analog filter circuit for a given requirement. At all stages, the signal being filtered is an electrical voltage or current, which is the direct analog of the physical quantity (example, a sound or video signal or transducer output) involved.

A digital filter uses a digital processor to perform numerical calculations on sampled values of the signal. The processor may be a general-purpose computer such as a PC, or a specialized DSP (Digital Signal Processor) chip. The analog input signal must first be sampled and digitized using an ADC. The resulting binary numbers, representing successive sampled values of the input signal, are transferred to the processor, which carries out numerical calculations on them. These calculations typically involve multiplying the input values by constants and adding the products together. If necessary, the results of these calculations, which now represent sampled values of the filtered signal, are output through a DAC (digital to analog converter) to convert the signal back to analog form. Note that in a digital filter, the signal is represented by a sequence of numbers, rather than a voltage or current.

The main advantages of digital filters over analog filters are listed below.

1. A digital filter is programmable, that is, its operation is determined by a program stored in the processor's memory. This means the digital filter can easily be changed without affecting the circuitry (hardware). An analog filter can only be changed by redesigning the filter circuit.
2. Digital filters are easily designed, tested and implemented on a general-purpose computer or workstation.
3. The characteristics of analog filter circuits (particularly those containing active components) are subject to drift and are dependent on temperature. Digital filters do not suffer from these problems, and so are extremely stable with respect to both time and temperature.
4. Unlike their analog counterparts, digital filters can handle low frequency signals accurately. As the speed of DSP technology continues to increase, digital filters are being applied to high frequency signals in the RF (radio frequency) domain, which in the past was the exclusive preserve of analog technology.
5. Digital filters are very much more versatile in their ability to process signals in a variety of ways; this includes the ability of some types of digital filter to adapt to changes in the characteristics of the signal.

Fast DSP processors can handle complex combinations of filters in parallel or cascade (series), making the hardware requirements relatively simple and compact in comparison with the equivalent analog circuitry.



CHAPTER2

Introduction to Signals and Systems

2.1 INTRODUCTION TO MODELING

This book discusses signals and systems related to Engineering. It focuses on the modeling of physical signals and systems by mathematical functions, and the solution of such mathematical functions, when the system is excited by such signals.

2.1.1 Signals

A signal is defined as a function of one or more variables which conveys information. A signal is a physical quantity that varies with time in general, or any other independent variable. It can be dependent on one or more independent variables. Dimension of a signal may be defined based on the number of independent variables.

Any variables which does not convey information is called Noise. Noise is a random phenomenon in which physical parameters are time-variant. Unlike a signal, noise is usually does not carry useful information and is almost always considered undesirable. Some examples include channel noise in communication systems, transformer humming in electrical engineering and moving artifacts in biological systems.

2.1.2 One-dimensional Signal

When a function depends on a single independent variable to represent the signal, it is said to be a one-dimensional signal.

The ECG signal and speech signal shown in Fig. 2.1(a) and 2.1(b) respectively are examples of one-dimensional signals where the independent variable is time. The magnitude of the signals is dependent variable.

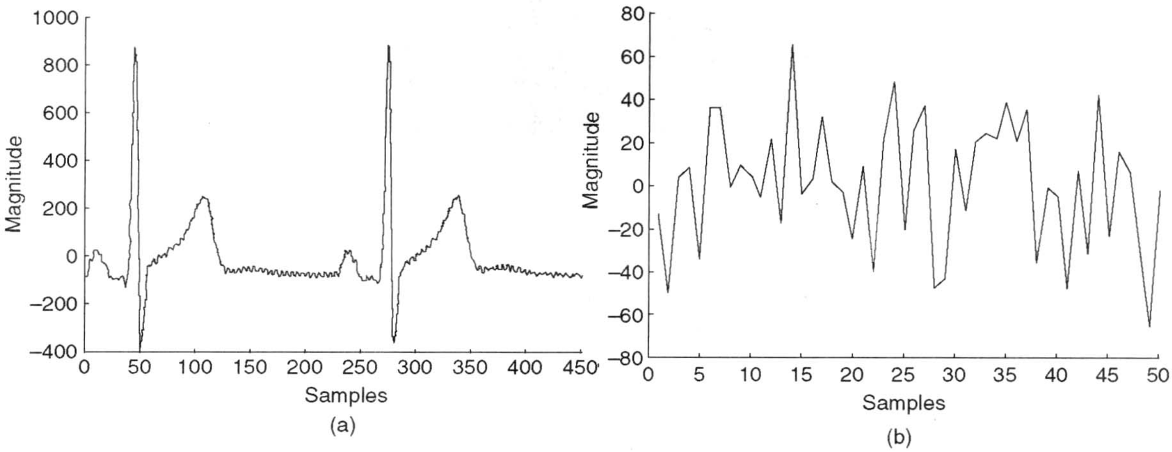


Fig. 2.1 One-dimensional Signal

(a) ECG Signal (b) Speech Signal

2.1.3 Two-dimensional Signal

When a function depends on two independent variables to represent the signal, it is said to be a two-dimensional signal. For example, photograph shown in Fig. 2.2 is an example of two-dimensional signal wherein the two independent variables are the two spatial coordinates which are usually denoted by x and y.

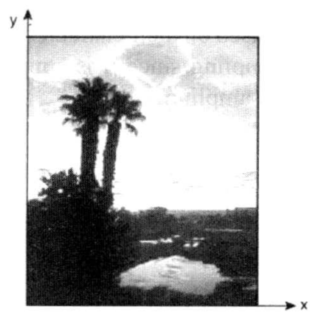


Fig. 2.2 Two-dimensional Photograph

2.1.4 Multi-dimensional Signal

When a function depends on more than one independent variables to represent the signal, it is said to be a multi-dimensional signal. For example, space missile shown in Fig. 2.3 is an example of three-dimensional image.

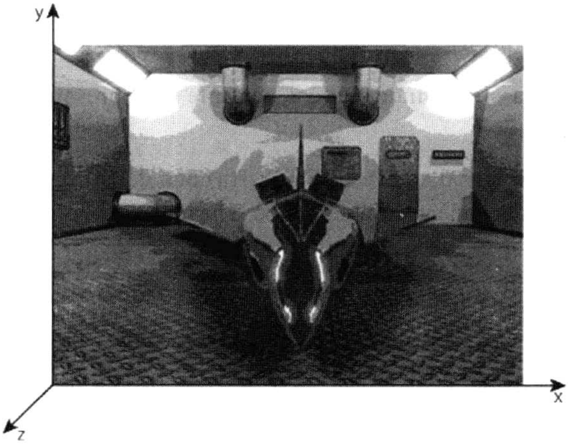


Fig. 2.3 3D-Space Missile

Definition

Input signal A signal that enters a system from an external source is referred to as an input signal. For example, the voltage from a function generator, electrocardiogram from heart, temperature from the human body, etc.

Output signal A signal produced by the system (may or may not be) in response to the input signal is called the output signal. For example, displacement due to force, output voltage from an amplifier, sinusoidal signal from an oscillator, etc.

2.1.5 Sampling

Sampling is a process by which a continuous-time signal (continuous with respect to time) is converted into a sequence of discrete samples, with each sample representing the amplitude of the signal at a particular instant of time. The sampling can be either uniform or non-uniform sampling.

In uniform sampling, the space between any two samples is fixed throughout the signal under consideration. A uniform sampling is illustrated in Fig. 2.4. In nonuniform sampling, the space between any two samples varies throughout the signal under consideration based on their characteristics like frequency, etc. In general, uniform sampling is preferred over nonuniform sampling since it is simple to analyze and easy to implement. The hardware complexity is also low in uniform sampling.

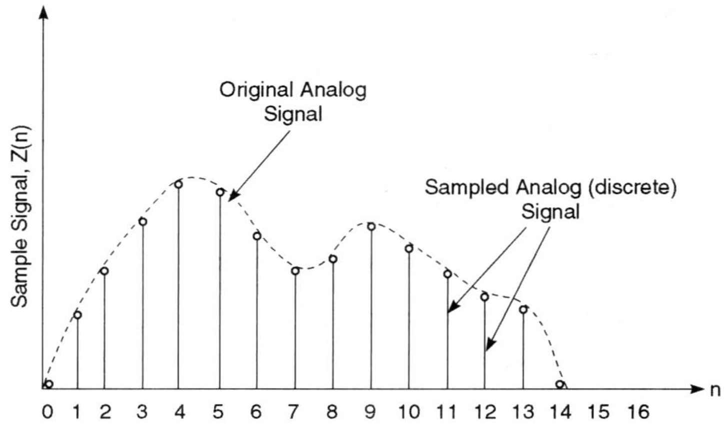


Fig. 2.4 Uniform Sampling of signal

2.1.6 Quantization

Quantization is a process by which each sample produced by the sampling circuit to the nearest level is selected from a finite number of discrete amplitude level as illustrated in Fig. 2.5.

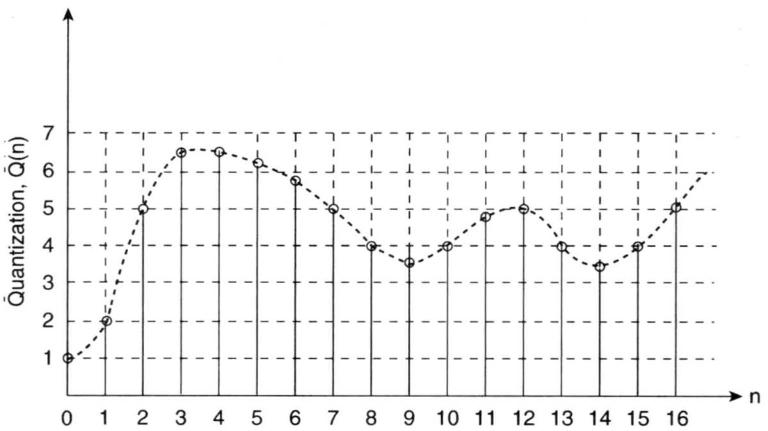


Fig. 2.5 Quantization of Signal

2.1.7 Coding

Coding is needed in order to represent each quantized sample by a binary number '0' or '1'. The '0' represents the "low" state or logical '0', and '1' represents the "high" state or logical '1'. The encoded version of quantized signal of Fig. 2.5 is shown in Table 2.1.

Table 2.1 3-bit Quantization and its Binary Representation

n	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Q(n)	1	2	5	7	7	7	6	5	4	4	4	5	5	4	4	4	5
Binary	001	010	101	111	111	111	110	101	100	100	100	101	101	100	100	100	101

2.2 CLASSIFICATION OF SIGNALS

Signals are classified based on their fundamental properties. They are:

1. Continuous-time signal and Discrete-time signal
2. Periodic signal and Aperiodic signal
3. Even signal and Odd signal
4. Deterministic signal and Random signal
5. Energy signal and Power signal

2.2.1 Continuous-time Signal and Discrete-time Signal

Signal can be represented either by continuous or discrete values.

Continuous-time signal A signal $x(t)$ is said to be a continuous-time signal if it is defined for all time t. The amplitude of the signal varies continuously with time. In general, all signals by nature are continuous-time signals.

The speech signal is a continuous-time signal, that is, conversation between persons is continuous with respect to time (Fig. 2.6a).

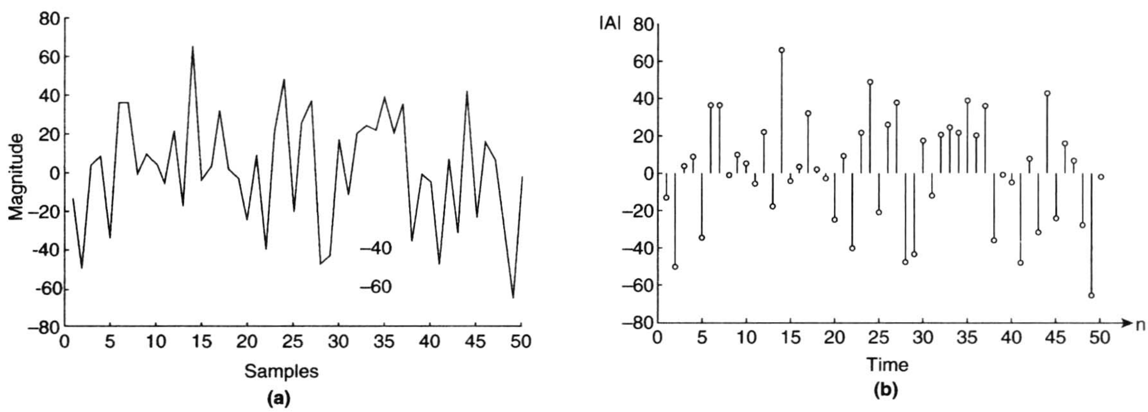


Fig. 2.6 (a) Continuous-time Signal Representation of Speech Signal
(b) Discrete-time Signal Representation of Speech Signal