

The book cover features a diagonal split. The upper right portion is dark blue with glowing orange and blue circuit traces. The lower left portion is a vibrant red with glowing white and red circuit traces. The title 'DIGITAL SIGNAL PROCESSING' is written in large, bold, white, sans-serif capital letters across the center, overlapping both background sections.

DIGITAL SIGNAL PROCESSING

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Digital Signal Processing

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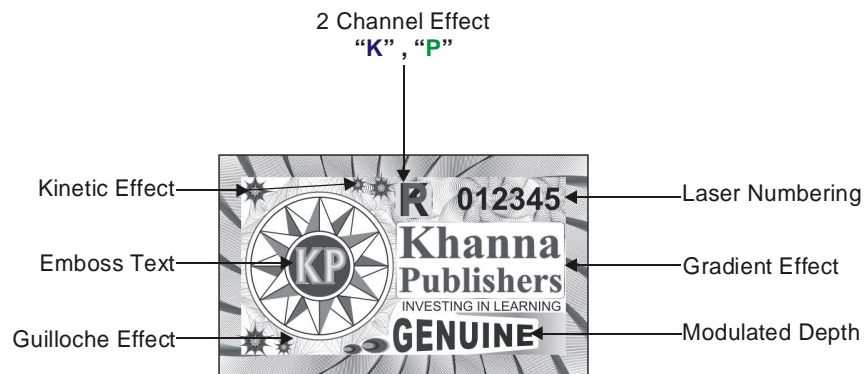
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Preface

The great advancements in the design of microchips, digital systems, and computer hardware over the past 40 years have given birth to digital signal processing (DSP) which has grown over the years into a ubiquitous, multifaceted, and indispensable subject of study. As such DSP has been applied in most disciplines ranging from engineering to economics and from astronomy to molecular biology. Consequently, it would take a multivolume encyclopedia to cover all the facets, aspects, and ramifications of DSP, and such a treatise would require many authors. This textbook focuses instead on the fundamentals of DSP, namely, on the representation of signals by mathematical models and on the processing of signals by discrete-time systems. Various types of processing are possible for signals but the processing of interest in this volume is almost always linear and it typically involves reshaping, transforming, or manipulating the frequency spectrum of the signal of interest.

This author considers the processing of continuous- and discrete-time signals to be different facets of one and the same subject of study without a clear demarcation where the processing of continuous-time signals by analog systems ends and the processing of discrete-time signals by digital systems begins. Discrete-time signals sometimes exist as distinct entities that are not derived from or related to corresponding continuous-time signals. The processing of such a signal would result in a transformed discrete-time signal, which would be, presumably, an enhanced or in some way more desirable version of the original signal. Obviously, reference to an underlying continuous time signal would be irrelevant in such a case. However, more often than not discrete-time signals are derived from corresponding continuous-time signals and, as a result, they inherit the spectral characteristics of the latter. Discrete-time signals of this type are often processed by digital systems and after that they are converted back to continuous-time signals. A case in point can be found in the recording industry where music is first sampled to generate a discrete-time signal which is then recorded on a CD. When the CD is played back, the discrete-time signal is converted into a continuous-time signal. In order to preserve the spectrum of the underlying continuous-time signal, e.g., that delightful piece of music, through this series of signal manipulations, special attention must be paid to the spectral relationships that exist between continuous- and discrete-time signals.

In the past signal processing appeared in various concepts in more traditional courses like telecommunications, control, circuit theory, and in instrumentation. The signal processing done was analog and discrete components were used to achieve the various objectives. However, in the later part of the 20th century we saw the introduction of computers and their fast and tremendous growth. In the late 1960s and early 1970s a number of researchers resorted to modeling and simulation of various concepts in their research endeavors, using digital computers, in order to determine performance and optimize their design. It is these endeavors that led to the development of many digital signal processing algorithms that we know today. With the rapid growth of computing power in terms of speed and memory capacity a number of researchers wanted to obtain their results from near real-time to real time. This saw the development of processors and I/O devices that were dedicated to real-time data processing though initially at lower speeds they are currently capable of processing high speed data including video signals. The many algorithms that were developed in the research

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activities combined with software and hardware that was developed for processing by industry ushered in a new course into the Universities curricula; Digital Signal Processing.

For many years the course Digital Signal processing was offered as a postgraduate course with students required to have a background in telecommunications (spectral analysis), circuit theory and of course Mathematics. The course provided the foundation to do more advanced research in the field. Though this was very useful it did not provide all the necessary background that many industries required; to write efficient programs and to develop applications. In many institutions a simplified version of the postgraduate course has filtered into the undergraduate programme. In many cases that we have examined this course is a simplified version of the postgraduate course, it is very theoretical and does not pass the necessary tools to students that industry requires. This book is an attempt to bridge the gap.

The book can serve as a text for undergraduate or graduate courses and various scenarios are possible depending on the background preparation of the class and the curriculum of the institution.

New Delhi

—Rishabh Anand

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CHAPTER

1

INTRODUCTION TO DIGITAL SIGNAL PROCESSING (DSP)

1.1 INTRODUCTION

Digital Signal Processing (DSP) is an area of science and technology that has developed rapidly over the past few decades. The techniques and applications of DSP are as old as Newton and Gauss and as new as Digital Computers and Integrated Circuits (ICs). The rapid development of DSP is a result of the significant advances in Digital Computer Technology and IC fabrication.

DSP is concerned with the representation of signals by sequences of numbers or symbols and processing of these sequences. Processing means modification of sequences into a form which is in some sense more desirable.

In another words, DSP is a mathematical manipulation of discrete-time signals to get more desirable properties of the signal, such as less noise or distortion.

The classical numerical analysis formulae such as those used for interpolation, differentiation and integration are also DSP algorithms.

DSP finds application in various fields such as speech communication, data communication, image processing, radar engineering, seismology, sonar engineering, biomedical engineering, acoustics, nuclear science and many others.

DSP can be applied to one dimensional signals as well as multidimensional signals. Example of one dimensional signal is speech and example of two-dimensional signal is image. Many picture processing applications require the use of two dimensional signal processing techniques. Two-dimensional signal processing includes X-ray enhancement, analysis of aerial photographs (these photographs are necessary for detection of forest fire or crop damage), analysis of satellite weather photographs etc. Analysis of seismic data is required in oil exploration, earth quake measurements and monitoring of nuclear tests. These utilize multidimensional signal processing techniques. The impact of DSP techniques will undoubtedly promote revolutionary advances in many fields of application. A notable example is telephony where digital techniques dramatically increased economy and flexibility in implementing switching and transmission systems.

1.2 APPLICATIONS OF DIGITAL SIGNAL PROCESSING

There are a variety of application areas of digital signal processing because of the availability of high resolution spectral analysis. It requires high speed processor to implement the Fast Fourier Transform (FFT). Some of these areas are :

1. Speech processing
2. Image processing
3. Radar signal processing

4. Digital communications
5. Spectral analysis
6. Sonar signal processing

Many of the above applications are discussed in Chapter 13.

Some of the other applications of digital signal processing are in :

- (a) Transmission lines
- (b) Advanced optical fiber communication
- (c) Analysis of sound and vibration signals
- (d) Implementation of speech recognition algorithms
- (e) Very Large Scale Integration (VLSI) technology
- (f) Telecommunication networks
- (g) Microprocessor systems
- (h) Satellite communications
- (i) Telephony transmission
- (j) Aviation
- (k) Astronomy
- (l) Industrial noise control
- (m) New DSP algorithms and many more.

Speech Processing: Speech is a one dimensional signal. Digital processing of speech is applied to a wide range of speech problems such as speech spectrum analysis, channel vocoders (voice coders) etc. DSP is applied to speech coding, speech enhancement, speech analysis and synthesis, speech recognition and speaker recognition.

Image Processing: Any two-dimensional pattern is called an image. Digital processing of images requires two-dimensional DSP tools such as Discrete Fourier Transform (DFT), Fast Fourier Transform (FFT) algorithms and z -transforms. Processing of electrical signals extracted from images by digital techniques include image formation and recording, image compression, image restoration, image reconstruction and image enhancement.

Radar Signal Processing: Radar stands for “Radio Detection and Ranging”. Improvement in signal processing is possible by digital technology. Development of DSP has led to greater sophistication of radar tracking algorithms. Radar systems consist of transmit-receive antenna, digital processing system and control unit.

Digital Communications: Application of DSP in digital communication specially tele-communications comprises of digital transmission using PCM, digital switching using time division multiplexing (TDM), echo control and digital tape recorders. DSP in telecommunication systems are found to be cost effective due to availability of medium and large scale digital ICs. These ICs have desirable properties such as small size, low cost, low power, immunity to noise and reliability.

Spectral Analysis: Frequency-domain analysis is easily and effectively possible in digital signal processing using Fast Fourier Transform (FFT) algorithms. These algorithms reduce computational complexity and also reduce the computational time.

Sonar Signal Processing: Sonar stands for “Sound Navigation and Ranging”. Sonar is used to determine the range, velocity and direction of targets that are remote from the observer. Sonar uses sound waves at lower frequencies to detect objects under water.

DSP can be used to process sonar signals, for the purpose of navigation and ranging.

1.3 SIGNAL

A signal can be defined as a function of one or more independent variable(s) which conveys information. Independent variables may be time, space etc. depends on the type of signals.

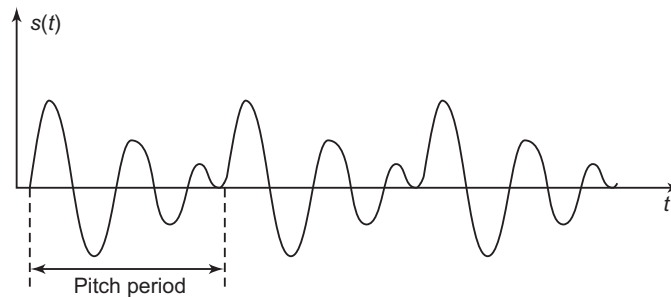


Fig. 1.1 Speech signals.

Example of signals are speech signals, picture, Electrocardiogram (ECG) signal etc. A speech signal is represented mathematically as a function of time and a picture signal is represented as a brightness function of two spatial variables.

1.4 CLASSIFICATION OF SIGNALS

Any investigation in signal processing is started with a classification of signals involved in the specific application. Signals can be classified in the following classes :

1. Multichannel and Multidimensional signals
2. Continuous-time and Discrete-time signals
3. Analog and Digital signals
4. Deterministic and Random signals
5. Energy and Power signals
6. Periodic and Non-periodic signals.

Now we will discuss these in detail in subsequent sections.

1.4.1 Multichannel and Multidimensional Signals

Multichannel Signals: Signals which are generated by multiple sources or multiple sensors are called Multichannel signals. These signals are represented by vector

$$s(t) = \begin{bmatrix} s_1(t) \\ s_2(t) \\ s_3(t) \end{bmatrix}$$

Above signal represents a 3-channel signal. In electrocardiography, 3-lead and 12-lead electrocardiograph is often used in practice, which results in 3-channel and 12-channel signals, respectively.

Multidimensional Signal: A signal is called multidimensional signal if it is a function of M independent variables. For example : *Speech signal* is a one dimensional signal because amplitude of signal depends upon single independent variable, namely, time. *TV Picture Signal* : A B/W picture signal is an example of 2-dimensional signal because brightness of the signal at each point is a function of two spatial independent variable, namely, x and y . Variables x and y are width and height of the picture element.

A coloured picture signal is an example of 3-dimensional signal because brightness of the signal at each point is a function of three independent variables, namely, x , y and time (t).

1.4.2 Continuous-time and Discrete-time Signals

Continuous-time Signals: A signal that varies continuously with time is called *continuous-time signal*. These are defined for every value of independent variable, namely, time. For example *speech signal and temperature* of the room are continuous-time signals. Continuous-time signal is shown in Fig. 1.2.

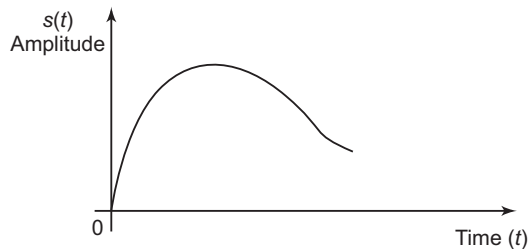


Fig. 1.2 Continuous-time signal.

Discrete-time Signal: Discrete-time signals are signals which are defined at discrete times (Fig. 1.3). These are represented by sequences of numbers. For example : Rail traffic signal is a discrete-time signal.

Discrete-time signals can be recovered by periodic sampling of continuous-time signals. Fig. 1.3 illustrates the discrete-time signal.

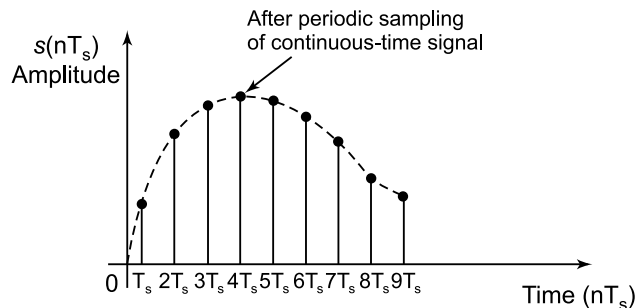


Fig. 1.3 Discrete-time signal.

1.4.3 Analog and Digital Signals

Analog Signals: Analog signals are signals whose both dependent variable and independent variable(s) are continuous in nature. Analog signals arise when a physical waveform is converted into an electrical signal. This conversion is performed by means of a transducer. For example : Telephone speech signals, TV signals etc., are very common types of analog signal.

Telephone Speech Signals. A telephone message comprises of speech sounds having vowels and consonants. These sounds produce an audio signal. These sound waves are converted into analog electrical signals by means of a transducer (microphone). Transducer is a device which converts non-electrical quantity into electrical signals. Example : Microphone. Continuous-amplitude, continuous-time signals are called *analog signals*. Analog signal is shown in Fig. 1.1.

Digital Signals: Digital signals are signals whose both dependent variable and independent variables are discrete in nature. Digital signals comprise of pulses occurring at

discrete intervals of time. Telegraph and teleprinter signals are the example of digital signals. Fig. 1.4 illustrates a telegraph signal.

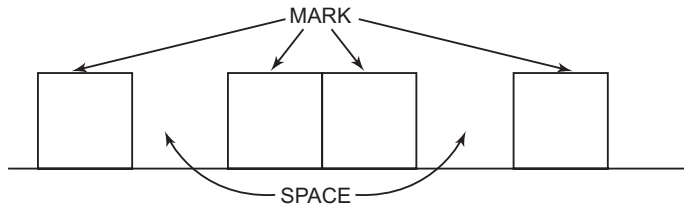


Fig. 1.4 Telegraph signal (Digital signal).

1.4.4 Deterministic and Random Signals

Deterministic Signals. A deterministic signal is one which has no uncertainty with respect to its value at any value of independent variable, namely, time. For example : Rectangular pulse given by Eqn. (1.1) is a deterministic signal. Fig. 1.5 and Fig. 1.6 illustrate rectangular pulse and cosine signal respectively, both are the example of deterministic signal.

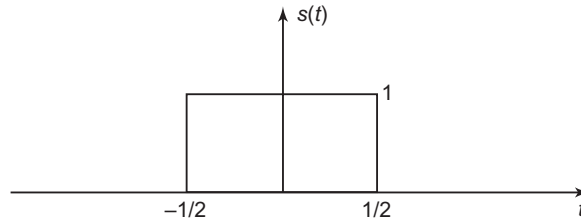


Fig. 1.5 Rectangular pulse.

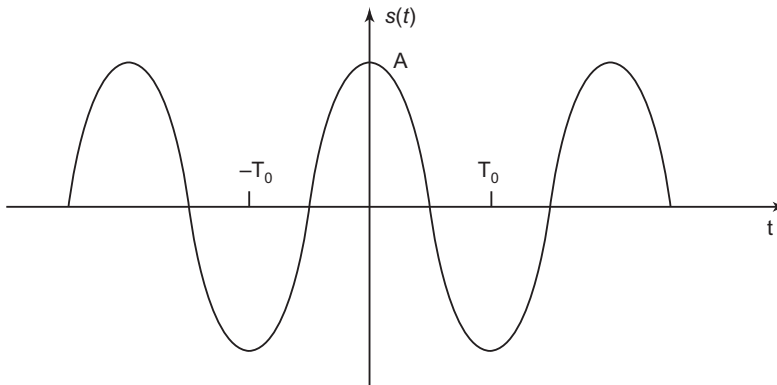


Fig. 1.6 Cosine signal.

$$s(t) = \begin{cases} 1, & |t| < \frac{1}{2} \\ 0, & \text{otherwise} \end{cases} \quad \dots(1.1)$$

Another example of deterministic signal is sinusoidal signals such as sine waves and cosine waves as given in Eqn. (1.2).

$$s(t) = A \cos \omega t, \quad -\infty < t < \infty \quad \dots(1.2)$$

Random Signal: A random signal is a signal which has some degree of uncertainty with respect to its value at any value of independent variable namely, time. For example : Thermal agitation noise in conductors is a random signal.

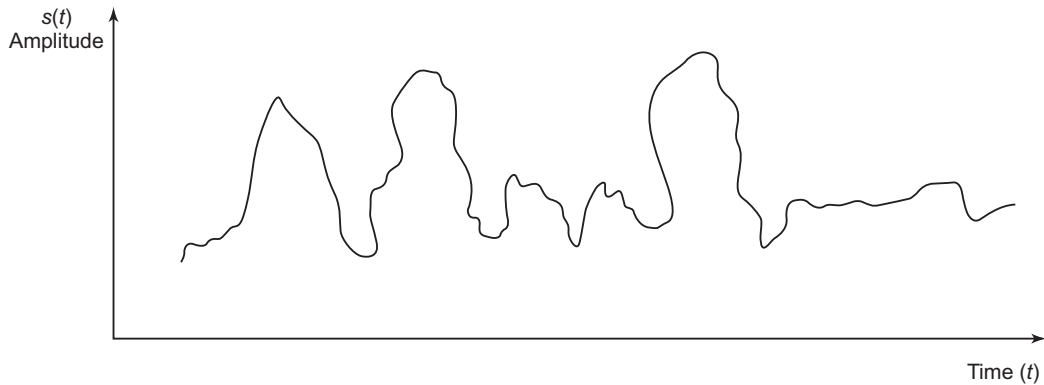


Fig. 1.7 Random signal.

1.4.5 Energy and Power Signals

Energy Signals: A signal is called energy signal if and only if its total energy is finite. For example : Rectangular pulse is an energy signal.

Power Signal: A signal is called power signal if and only if its average power is finite. For example : Sinusoidal waves are power signals.

The energy signals have zero average power and power signals have infinite energy. It means both signals are mutually exclusive.

1.4.6 Periodic and Non-periodic Signals

Periodic Signals: A signal which repeats its waveform after a fixed period of time is called as a *periodic signal*. This fixed time is called *Time-period* (T_0).

In other words, a signal which satisfies the condition $s(t) = s(t + T_0)$ for all t is called periodic signal. For example : sinusoidal signals are example of periodic signal.

Non-periodic Signal: A signal which does not satisfy above condition is called *non-periodic signals*.

Unit Rectangular pulse is the example of non-periodic signal.

Usually, periodic signals and random signals are power signals and deterministic signals and non-periodic signals are energy signals.

1.5 SIGNAL PROCESSING SYSTEMS

A system responds to particular signals by producing other signals having some desired behaviour.

Signal processing systems are of two types depending on the type of signal to be processed.

1. Continuous-time Systems.
2. Discrete-time Systems.

1.5.1 Continuous-time Systems

Continuous-time systems are the systems for which both input and output are continuous-time signals. $H(s)$ is the transfer function of a continuous-time system. Fig. 1.8 illustrates the block diagram of a continuous-time system.

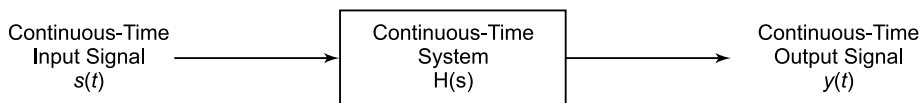


Fig. 1.8 Block diagram of continuous-time system.

An example of continuous-time system is an analog filter which is used to reduce the noise corrupting a message signal.

1.5.2 Discrete-time Systems

Discrete-time systems are systems for which both the input and output are discrete-time signals. $H(z)$ is the transfer function of a discrete-time system. Fig. 1.9 illustrates the block diagram of a discrete-time system.

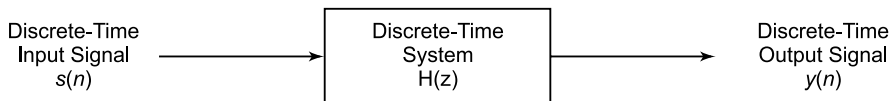


Fig. 1.9 Block diagram of discrete-time system.

An example of a discrete-time system is a digital computer.

1.6 SIGNAL PROCESSING

Changing the basic nature of signal to obtain the desired shaping of the input signal is called *signal processing*. Signal processing is concerned with the representation, transformation, and manipulation of signals and the information they contain.

Signal processing is of two types depending upon the type of signal to be processed.

1. Analog Signal Processing (ASP).
2. Digital Signal Processing (DSP).

1.6.1 Analog Signal Processing

In analog signal processing, continuous-amplitude continuous-time signals are processed. Various types of analog signals are processed through low pass filters, high pass filters, band pass filters and band reject filters to obtain the desired shaping of the input-signal. Another example of analog signal processing is the production of modulated carrier using High Frequency (HF) oscillator, and the modulating audio signal and a modulator. Fig. 1.10 illustrates the block diagram of an ASP system.

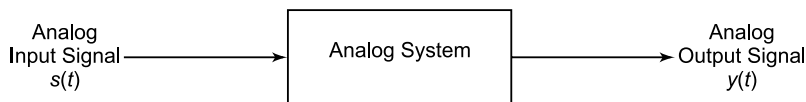


Fig. 1.10 Block diagram of ASP system.

1.6.2 Digital Signal Processing

Digital signal processing (DSP) is a numerical processing of signals on a digital computer or some other data processing machine. Fig. 1.11 illustrates the block diagram of DSP system.

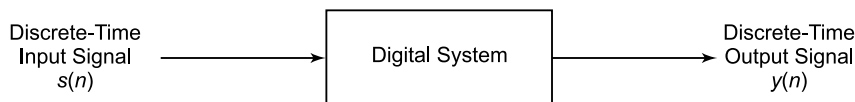


Fig. 1.11 Block diagram of DSP system.

A digital system such as digital computer takes input signal in discrete-time sequence form and converts it in discrete-time output sequence.

1.7 ADVANTAGES OF DIGITAL SIGNAL PROCESSING OVER ANALOG SIGNAL PROCESSING

Digital signal processing has following advantages :

1. Digital signal processing operations can be changed by changing the program in digital programmable system, *i.e.*, these are flexible systems.
2. Better control of accuracy in digital systems compared to analog systems.
3. Digital signals are easily stored on magnetic media such as magnetic tape without loss of quality of reproduction of signal.
4. Digital signals can be processed off line, *i.e.*, these are easily transported.
5. Sophisticated signal processing algorithms can be implemented by DSP method.
6. Digital circuits are less sensitive to tolerances of component values.
7. Digital systems are independent of temperature, ageing and other external parameters.
8. Digital circuits can be reproduced easily in large quantities at comparatively lower cost.
9. Cost of processing per signal in DSP is reduced by time-sharing of given processor among a number of signals.
10. Processor characteristics during processing, as in adaptive filters can be easily adjusted in digital implementation.
11. Digital system can be cascaded without any loading problems.

1.8 ELEMENTS OF DIGITAL SIGNAL PROCESSING SYSTEM

Majority of the signals encountered in science are analog in nature. In analog signals, both dependent variable and independent variable(s) are continuous. Such signals may be processed directly by analog systems (*i.e.*, analog filters) for the purpose of changing their characteristics or extracting some desired information.

Analog signals can also be processed digitally using DSP techniques. To process analog signals digitally, an interface between the analog signal and digital processor is needed. This interface is termed as analog-to-digital converter. The output of the analog-to-digital converter is a digital signal. This digital signal is appropriate for digital processor.

The digital signal processor may be a large programmable digital computer or a small microprocessor.

In some applications such as in speech communication. We require digital signal in analog form at the receiver end. Here we need another interface, called digital-to-analog converter. Fig. 1.12 illustrates the block diagram of a DSP system.

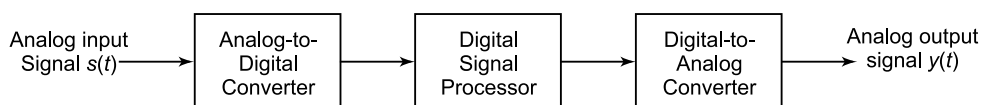


Fig. 1.12 Block diagram of a digital signal processing system.

REVIEW QUESTIONS

1. Define a signal. Give some examples of signals.
2. Give the classification of signals.
3. What is signal processing ? Differentiate between analog signal processing and digital signal processing.
4. What are the basic elements of digital signal processing system ?
5. What are the advantages of digital signal processing over analog signal processing ?
6. Differentiate multichannel and multidimensional signals. Give some examples of these signals.
7. What is the importance of DSP in various fields of engineering and technology ? Give a brief account of its applications.

DIGITAL SIGNAL PROCESSING

About the Book

Digital Signal Processing, First Edition, is a comprehensive textbook written with student-centered, pedagogically-driven approach, the text provides a self-contained introduction to the theory of digital signal processing offered by various universities in India as well as overseas. Considering the highly mathematical nature of this subject, more emphasis has been given on the problem-solving methodology. Considerable effort has been made to elucidate mathematical derivations in a step-by-step manner. Exercise problems with varied difficulty levels are given in the text to help students get an intuitive grasp on the subject.

This book with its lucid writing style and handy pedagogical features will prove to be a master text for engineering students and practitioners of Electronics and Communication engineering, Telecommunication engineering, Electronics and Instrumentation engineering, Electrical and Electronics engineering, Electronics and Computers engineering, Biomedical engineering and Medical Electronics engineering. This book will also be useful to AMIE and IETE students.

About the Author



Rishabh Anand is an eminent academician; plays versatile roles and responsibilities juggling between industry, research, publications and consultancy. He has completed his PostDoc (AI & ML) from Sao Paulo State University, Brazil, Ph.D.(Computer Science) from University of Bristol, United Kingdom, Degree of Master of Business Administration as MBA Management from International MBA Institute, Switzerland and Program Diploma in Innovation Management from International Business Management Institute, Germany. Also, he is the Reviewer/Editor for IJTESSS, IJISP,IJCAC, IJECME, IJICTE,JITR, IJMPA, IJTHI,IRJET,IJCRT and IJSDR. He is a prolific author with 34 Text and Reference books to his credit. He is also associated with various professional bodies like IEEE, LMCEGR, IAENG, Internet Society, IAOP, and IAOP. He is currently working in ITES MNC industry as a Global Service Delivery Manager with overall 15 years of experience. He is CDPT™,PMP®, PRINCE2®, DevOps-PM™, ITIL4®, CSM® & Kanban-ASC™ Certified Professional.



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