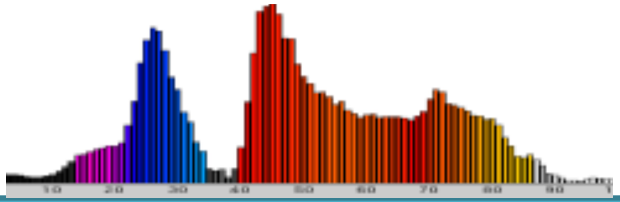
Department of Electrical an Engineering



EE416
Digital Signal Processing (DSP)

Asst. Prof. Dr. Ayad A. ABDULKAFI

2024-2025

Outlines

- **1. Digital Signals and Systems:**
- Digital Signals (DS): Common Types and Generation of DS.
- Basic concepts: linear, time variant, stable and casual discrete systems.
- Difference equations and impulse response. Input / output relations in Z- domains.
- Discrete convolution techniques: tabular method, Z- transform method, matrix method, add overlap method.
- De-convolution methods': iterative method and Z-transform method, frequency response of discrete system.
- **2. Discrete Fourier transform (DFT):**
- Definition and properties of DFT,
- Matrix formulation of the DFT., Fast
- Fourier transform. (FFT)
- Decimation in time (DIT), decimation in frequency (DIF), arithmetic complexity.
- **3. Digital Filter Design**
- Classification and realization of discrete system (FIR and IIR systems),
- Review of analogue filter design (Butterworth and Chebyshev filter), IIR
- filter design using analogue filters and the bilinear transformation, filter
- transformation for IIR (LPF/ LPF, LPF/HPF, LPF/BPF, LPF/BSF). FIR
- filter design using windows, Rectangular, Bartlett, Hanning, Hamming and
- Blackman windows, LPF, HPF, BPF and BSF FIR filter..

References:

- Lecture Notes

- Digital Signal Processing: Fundamentals and Applications, Li Tan, Jiang Jean, Second Edition, Elsevier, (2013).

- Digital Signal. Processing. Principles, Algorithms, and Applications. Third Edition. John G. Proakis. Northeastern University. Dimitris G. Manolakis.

- Linear Systems and Signals, 2nd edition, by B.P. Lathi

- Signals and Systems, 2nd edition by Alan V. Oppenheim

Basic concepts of DSP

Digital Signal Processing (DSP) is an area of science and engineering that is applied in different fields. It is concerned with the numerical manipulations of signals and data in sampled form. It is applied in various applications such as noise filtering, speech and audio enhancement, biomedical signal processing, oil exploration, detection of nuclear explosions, and image processing. DSP provides another way to process the analog signals efficiently and easily. DSP techniques are now used to analyze and process signals and data arising in many areas of engineering, science, medicine, economics and the social sciences.

Using elementary operations as digital storage, delay, addition, subtraction and multiplication by constants, we can produce a wide variety of useful functions. For example, to extract a wanted signal from unwanted noise, to assess the frequencies presented in a signal.

The general-purpose computer can be used for illustrating DSP theory and application. However, if high-speed real-time signal processing is required, it may use special purpose digital hardware. Programmable microprocessors attached to a general-purpose host computer. Various terms are used to describe signals in the DSP environment.

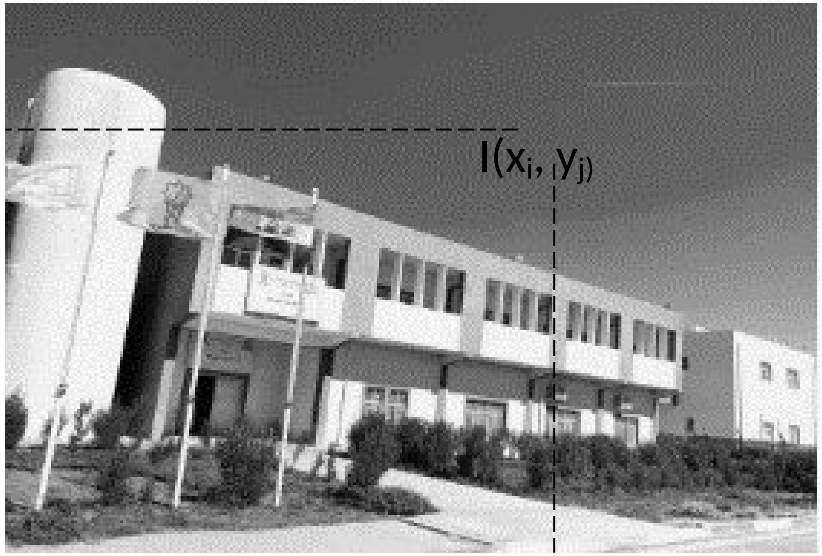
DSP takes real-world signals like voice, audio, video, temperature, pressure, or position that have been digitized and then mathematically manipulate them. A DSP is designed for performing mathematical functions like "add", "subtract", "multiply" and "divide" and other operations **very quickly.**

The scientific definition of Digital Signal Processing:

Digital: the word “digital” has different meaning based on the area of interest. For example, if the talking is about signals or data, the word digital represents as series of the digits 0 and 1 for the values of a physical quantity such as voltage. If the talking is about the digital system, this means systems with discrete inputs that produces outputs in the form of numbers.

Signal: is a function that carry information from one point to another point. It can be described by a function of one or more independent variable. The value of the function (i.e., the dependent variable) can be a real- valued scalar quantity, a complex-valued quantity, or perhaps a vector.

- Signals can have one independent variable such as audio, or two independent variable such as the image or three variable such as the video. In general, if the signal is a function of a single independent variable, the signal is called a **one-dimensional signal**. On the other hand, a signal is called ***M*-dimensional** if its value is a function of *M* independent variables.
- **For example**, the picture in the following figure is a 2D signal, because the brightness (or Intensity) is a function of two independent variables $I(x,y)$.



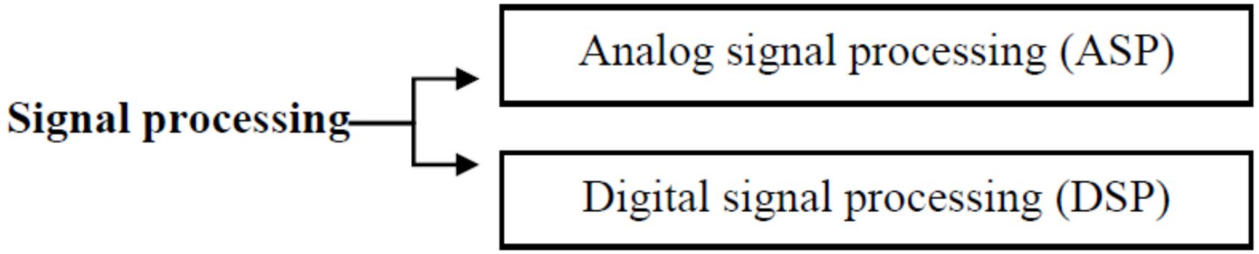
Introduction

In the same concepts, there are **single channel and multi- channel** signals based on the generation source of the signal. Because signals may be generated by multiple sources or multiple sensors. Such signals are represented in vector form. For $sk(t)$ is an electrical signal comes from kth sensor as a function of time, where k=1,2,3. Then p=3 signals is defined by a vector

$$S_3(t) = \begin{bmatrix} s_1(t) \\ s_2(t) \\ s_3(t) \end{bmatrix}$$

Note, the colour TV picture is a three-channel, three-dimensional signal, while the black-white TV picture can be written as $I(x,y,t)$ because the intensity is a function of time, so it is a three-dimensional signal.

Processing: the operations applied on the signal to either extraction of useful information to be used or to changing the signals characteristics. In general, **signal processing** can be done in two way to obtain the **processed signal** as shown:



As a conclusion: DSP extracts the desired signal from the undesired signal based on the general three topics:

1- Smoothing

2- Filtering.

3- Predicting.

These topics lead to estimate the wanted signal, which are called the estimation processes. The mathematical representation of the estimation process for the signal $x(t)$ to obtain the estimated signal $\hat{x}(t)$ is:

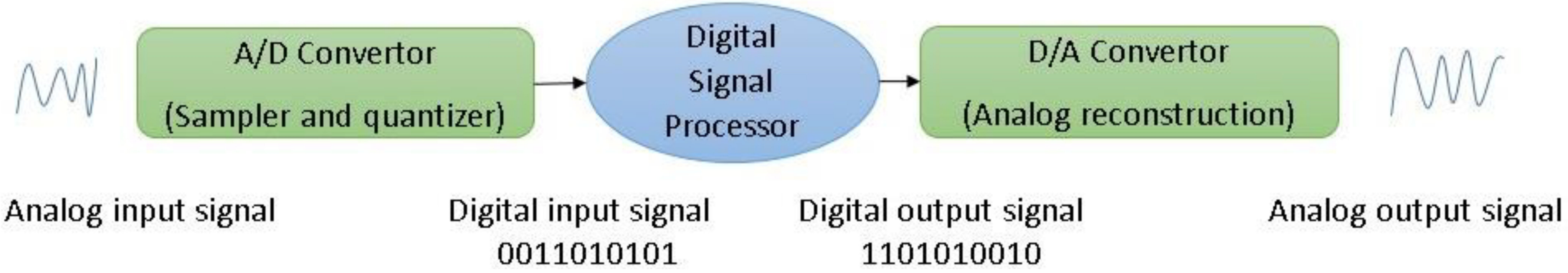
$$x(t) \xrightarrow{Estimation} \hat{x}(t)$$

Digital signal processing is implemented in three basic steps:

1. The analog signal is *digitized*, that is, it is *sampled* and each sample *quantized* to a finite number of bits. This process is implemented using an interface between the analog signal and the digital signal processor, which is called Analog- to- digital (A/D) convertor. It needs an analog filter prior ADC to determine frequency range of the signal before the sampling process.

2. The digitized samples are processed by a digital signal processor.

3. The resulting digital output samples are converted back into analog form using an analog reconstruct, that is the Digital- to- Analog convertor (D/A). An analog filter is also needed after the D/A to remove the sharp transitions from the output of the DAC.



Advantages of DSP over ASP

1. A digital programmable system allows flexibility in reconfiguring the digital signal processing operations simply by changing the program.

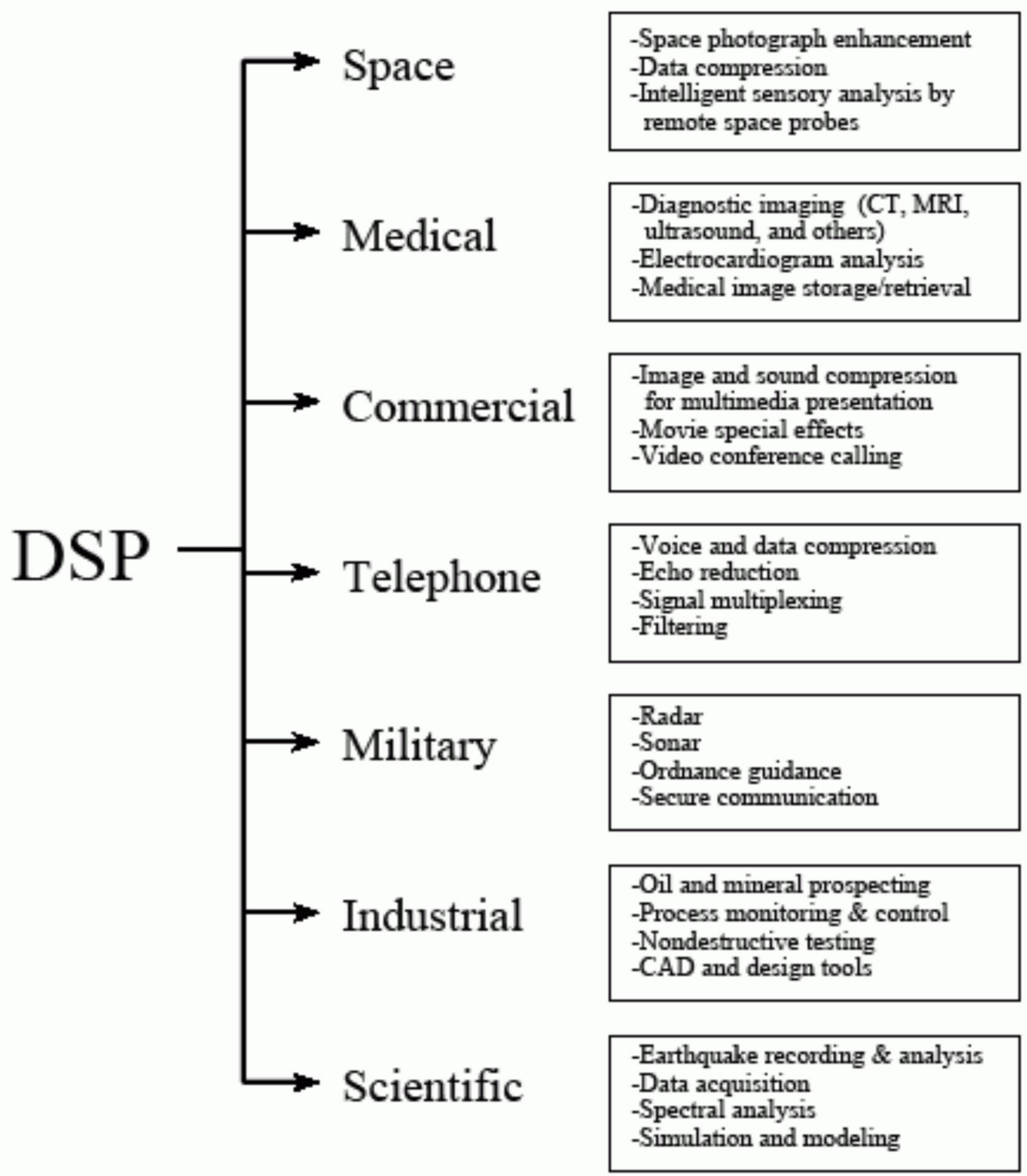
2. Signals and data are increasingly stored in digital computers and magnetic media, and transmitted from one place to another in digital form then processing them digitally. Moreover, the digital signal processor can be programmed to perform a variety of signal processing operations, such as filtering, spectrum estimation, and other DSP algorithms. Beside, depending on the speed and computational requirements of the application, the digital signal processor can be realized more easily.

3. Accuracy considerations also play an important role in determining the form of the signal processor. Tolerances in analog circuit components make it extremely difficult for the system designer to control the accuracy of an analog signal processing system .On the other hand, a digital system provides much better control of accuracy requirements. 5

However, digital implementation has its limitations, such as the limitation in the speed of operation of A/D converters and digital signal processors. For example, signals having extremely wide bandwidths require fast-sampling-rate A\D conversion and fast digital signal processor.

Some Applications of DSP

Magnetic cards, remote controls, digital entertainment systems, personal computers/networking, copiers/laser printers, new telecommunications systems (e.g., cellular phones, high-speed modems for Internet connection, video teleconferencing systems), health care apparatus, industrial control systems, automotive electronics, computerized billing/banking systems, and voice recognition/ synthesis. All above systems affect the way we live. Each of these applications has developed a deep DSP technology, with its own algorithms, mathematics, and specialized techniques. DSP has revolutionized many areas in science and engineering. In general, the most important applications in science and engineering that based on DSP can be divided as in the following figure:



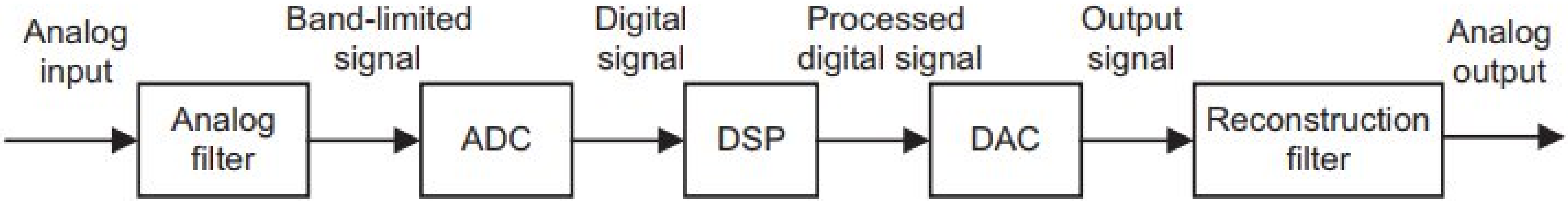


Fig . A typical DSP block diagram.

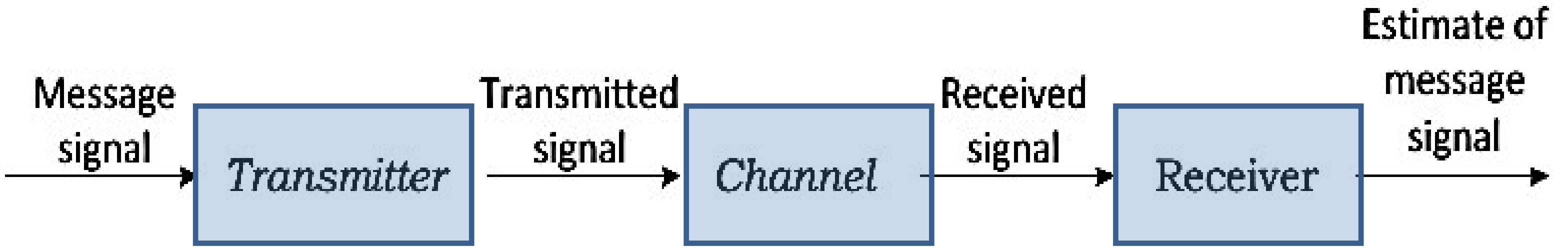
As shown in the diagram, the analog input signal, which is continuous in time and amplitude, is generally encountered in the world around us. Examples of such analog signals include current, voltage, temperature, pressure, and light intensity. Usually a transducer (sensor) is used to convert the nonelectrical signal to the analog electrical signal (voltage). This analog signal is fed to an analog filter, which is applied to limit the frequency range of analog signals prior to the sampling process. The purpose of filtering is to significantly attenuate aliasing distortion, which will be explained in the next chapter. The band-limited signal at the output of the analog filter is then sampled and converted via the ADC unit into the digital signal, which is discrete both in time and in amplitude. The DS processor then accepts the digital signal and processes the digital data according to DSP rules such as lowpass, high pass, and bandpass digital filtering, or other algorithms for different applications. Notice that the DS processor unit is a special type of digital computer and can be a general-purpose digital computer, a microprocessor, or an advanced microcontroller; furthermore, DSP rules can be implemented using software in general.

A signal is a set of data or information and can be defined as a function of one or more variables that conveys information on the nature of a physical phenomenon. Examples include a **telephon**e or a **television signal,** monthly sales of a corporation. Signals can either be one dimension e.g. speech or multidimensional e.g. image. For instance, in a RC circuit the signal may represent the voltage across the capacitor or the current flowing in the resistor. Mathematically, a signal is represented as a function of an independent variable ***t***. Usually ***t*** represents time. Thus, a signal is denoted by ***x(t).***

A system is defined as an entity that manipulates one or more signals to accomplish a function, thereby yielding new signals. It is a physical device that performs an operation on a signal, and it is characterized by the type of operation that performs on the signal. Such operations are referred to as signal processing.

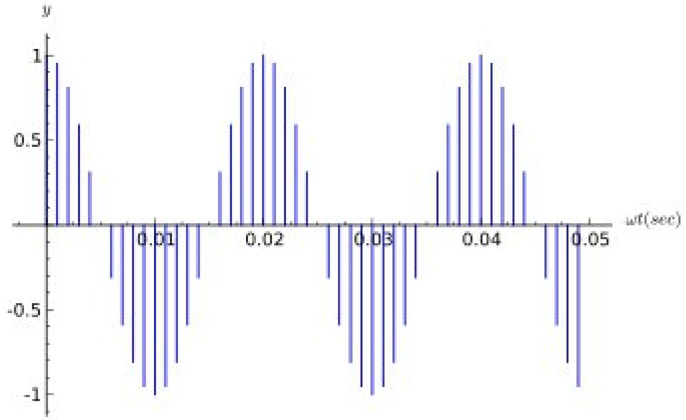
For example, in communication system, the input signal could be a speech signal or computer data, the system is made up of a transmitter, channel, and receiver, and the output is an estimate of the information contained in the original message as illustrate below.

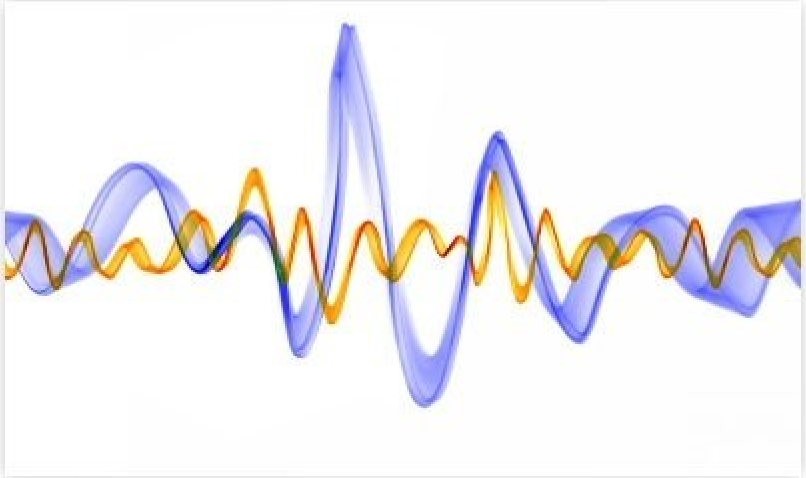
Communication System



Signals are classified according to their characteristics

1. Continuous time and discrete time signals
2. Analog and Digital Signals:
3. Real and Complex Signals:
4. Deterministic and random signals
5. Even and odd signals
6. Periodic and non periodic signals
7. Energy and power signals





Continuous-Time and Discrete-Time Signals:

A signal ***x(t)*** is a continuous-time signal if ***t*** is a continuous variable. By the term continuous signal we mean a real or complex function of time ***x(t),*** where the independent variable ***t*** is continuous. Several particularly important continuous signals are introduced later. These signals serve as basic building blocks from which other signals can be constructed. Therefore, the use of these signals allows us to examine the properties of systems.

If ***t*** is a discrete variable, that is, ***x(t)*** is defined at discrete times, then ***x(t)*** is a discrete-time signal. Since a discrete-time signal is defined at discrete times, a discrete-time signal is often identified as a sequence of numbers, denoted by {***x***,) or ***x[n],*** where ***n*** = integer. Illustrations of a continuous-time signal ***x(t)*** and of a discrete-time signal ***x[n]*** are shown in Fig. 1-1.

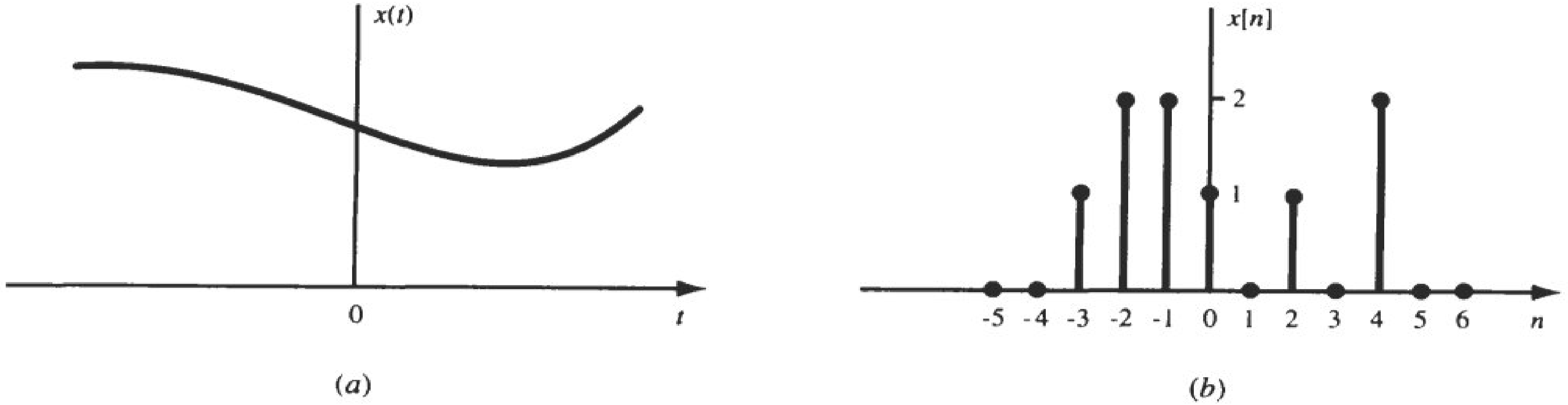


Fig. 1-1 Graphical representation of (a) continuous-time and (b)discrete-time signals.

Continuous-Time and Discrete-Time Signals:

A discrete-time signal ***x[n]*** may represent a phenomenon for which the independent variable is inherently discrete. For example, the weekly peak value of a particular economic indicator. On the other hand a discrete-time signal ***x[n]*** may be obtained by sampling a continuous-time signal ***x(t)*** such as

$$x(t_o), x(t_1), x(t_2), \ldots\ldots x(t_n), = x[0], x[1], x[2], \ldots. x[n] = x_0, x_1, x_2, \ldots\ldots x_n$$

$$\therefore \quad x_n = x[n] = x(t_n)$$

and x_n's are called samples and the time interval between them is called the sampling interval T_s. When the sampling intervals are equal (uniform sampling), then (See Fig. 1- 2)

$$x_n = x[n] = x(nT_s)$$

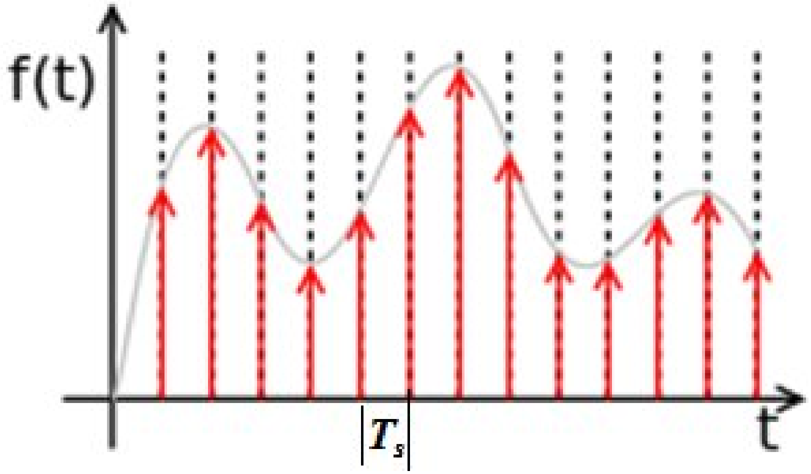


Fig. 1-2 Discrete sampled signal