Electrical and Computer Engineering Department
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Introduction

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A **signal**
is a function that conveys information about the behavior or attributes of some phenomenon.

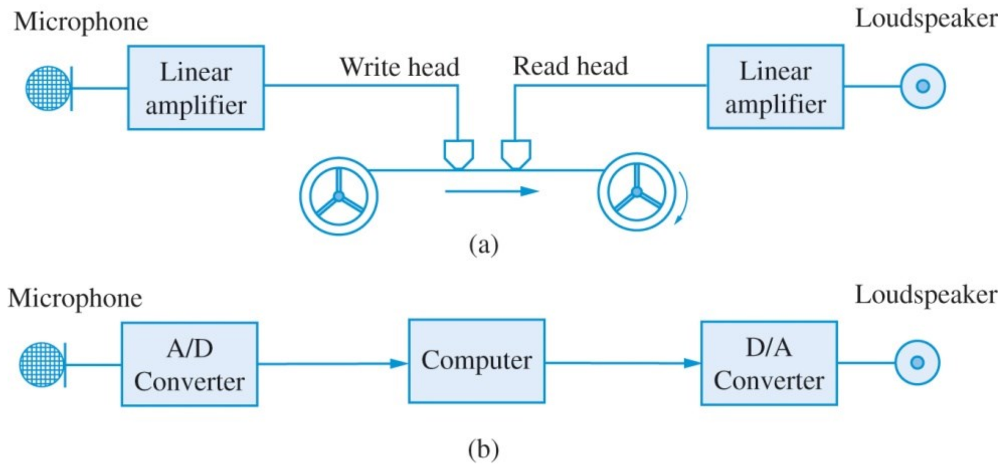
Signal processing
is a discipline concerned with the

- **acquisition**,
- **representation**,
- **manipulation**,
- and **transformation**

of signals.

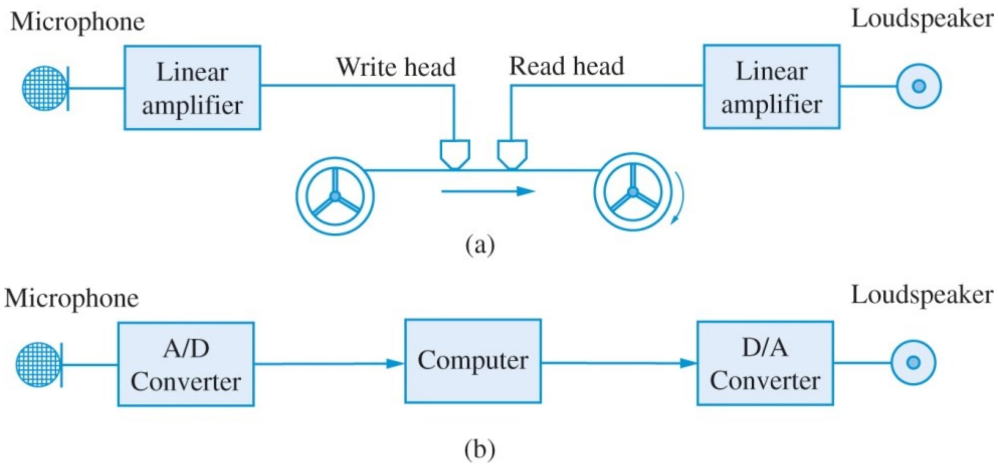
An analog audio recording system using magnetic tape

(a)



- Sound waves are picked up by a microphone and converted to a small analog voltage called the audio signal.
- The audio signal, which varies continuously to “mimic” the volume and frequency of the sound waves, is amplified and then converted to a magnetic field by the recording head.
- As the magnetic tape moves under the head, the intensity of the magnetic field is recorded (“stored”) on the tape.
- As the magnetic tape moves under the read head, the magnetic field on the tape is converted to an electrical signal, which is applied to a linear amplifier.
- The output of the amplifier goes to the speaker, which changes the amplified audio signal back to sound waves. The volume of the reproduced sound waves is controlled by the amplifier.

A digital recording system using a personal computer (b)



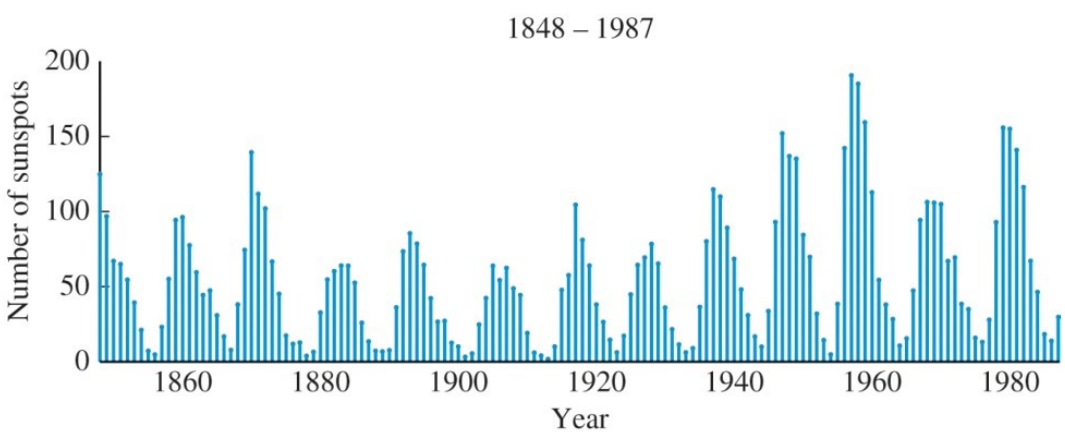
- The sound waves are converted to an electrical audio signal by the microphone. The audio signal is amplified to a usable level and is applied to an analog-to-digital converter.
- The amplified audio signal is converted into a series of numbers by the analog-to-digital converter.
- The numbers representing the audio signal can be stored or manipulated by software to enhance quality, reduce storage space, or add special effects.
- The digital data are converted into an analog electrical signal; this signal is then amplified and sent to the speaker to produce sound waves.

Mathematical representation of signals

- ***Continuous-time (analog) signals***
  The value *s(t)* are defined for every value of time *t*. The amplitude of a continuous-time signal may take any value from a continuous range of real numbers.

- ***Discrete-time signals***
  Are defined only at discrete times, that is, at a discrete set of values of the independent variable.

Discrete-time signal showing the annual mean sunspot number determined using reliable data collected during the 13 cycles from 1848 to 1987.



Sampling

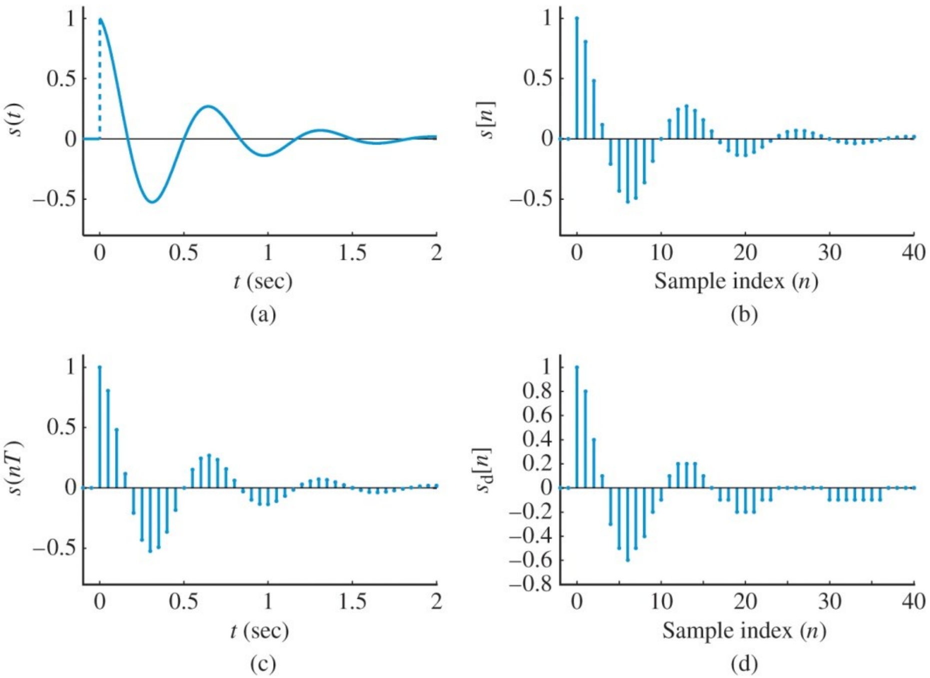
Most signals of practical interest arise as continuous-time signals. However, the use of digital signal processing technology requires a discrete-time signal representation. This is usually done by *sampling* a continuous-time signal at isolated, equally spaced points in time (periodic sampling). The result is a sequence of numbers defined by

$$s[n] \triangleq s(t)|_{t=nT} = s(nT),$$

where n is an integer $\{\ldots, -1, 0, 1, 2, 3, \ldots\}$ and T is the sampling period.

The quantity $F\text{s} \triangleq 1/T$, known as *sampling frequency* or *sampling rate*, provides the number of samples per second.

- ***Digital signal***
  A discrete-time signal $s[n]$ whose amplitude takes values from a finite set of K real numbers $\{a_1, a_2, \ldots, a_K\}$.



$$s(t) = \begin{cases} e^{-2t}\cos(3\pi t), & t \geq 0 \\ 0, & t < 0. \end{cases}$$

$$s[n] = s(nT) = \begin{cases} e^{-0.2n}\cos(0.3\pi n), & n \geq 0 \\ 0, & n < 0 \end{cases}$$

$T = 0.1$ s

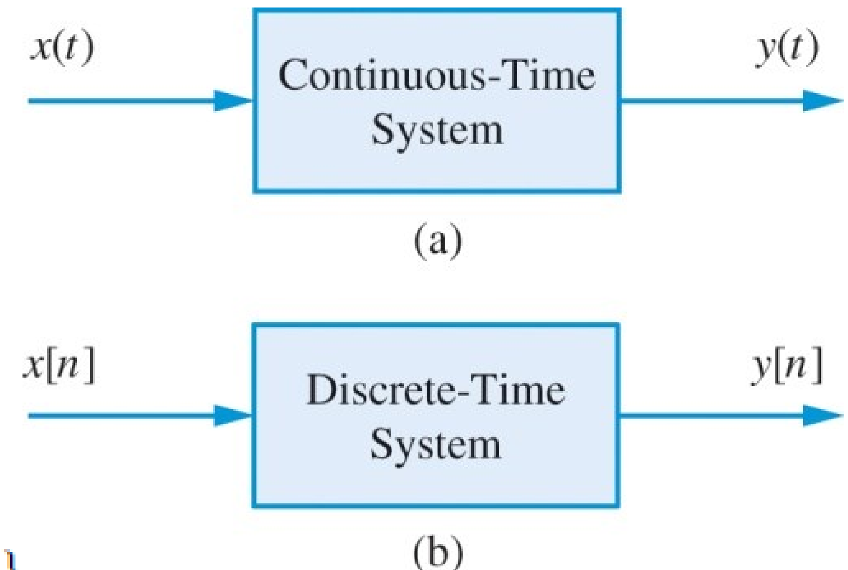
Plots illustrating the graphical representation of continuous-time signals (a), discrete-time signals (b) and (c), and digital signals (d).

Examples of typical signals

- **Music**
- **Speech**
- **Images**
- **Videos**
- **Electroencephalogram (EEG)**
- **Electrocardiogram (ECG)**
- **Motion**
- **Seismic**
- **…**

Systems

A process where a signal called *input* is transformed into another signal called *output*.



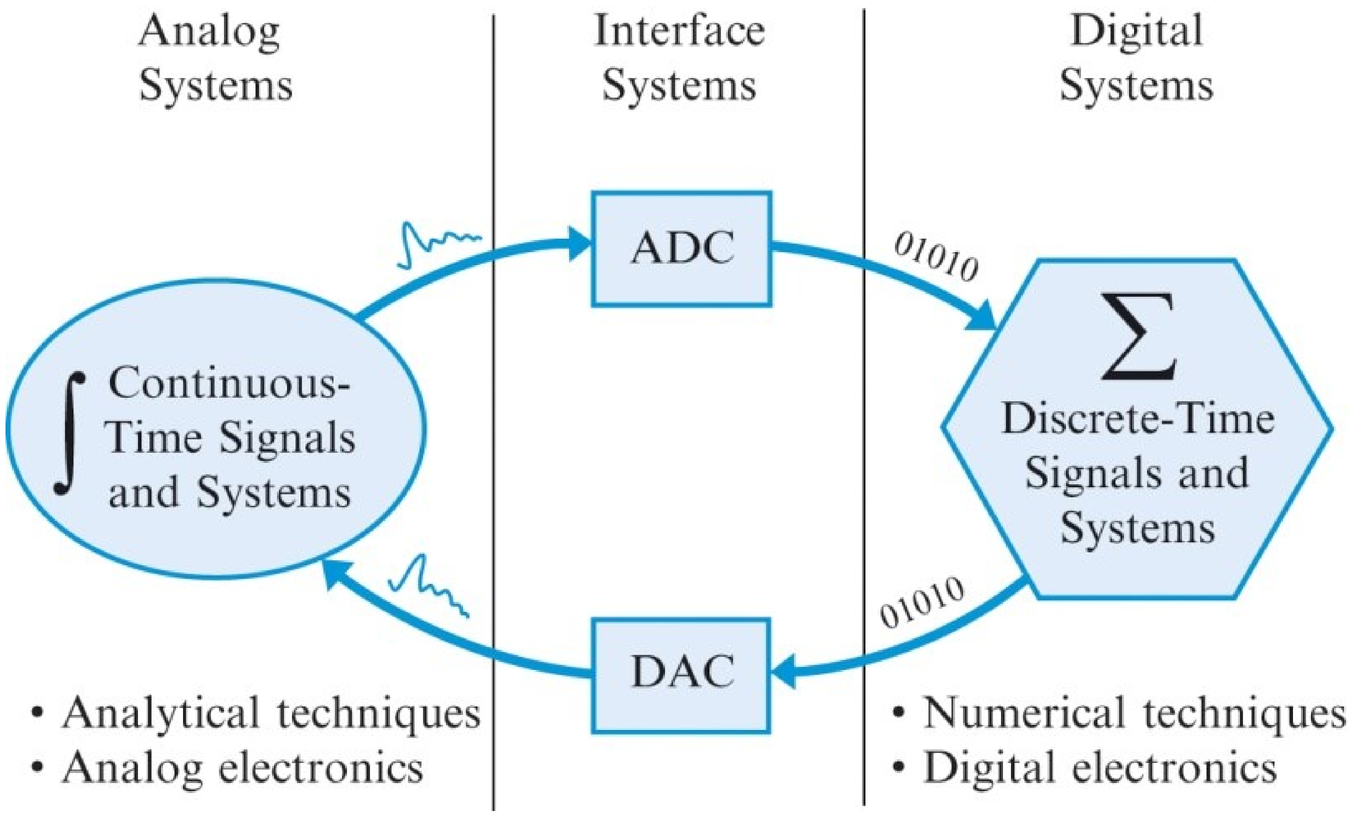
Continuous-time system:
A system which transforms a continuous-time input signal *x(t)* into a continuous-time output signal *y(t)*.

$$x(t) \xmapsto{\mathcal{H}} y(t) \quad \text{or} \quad y(t) = \mathcal{H}\{x(t)\},$$

***Discrete-time system*:**
A system that transforms a discrete-time input signal $x[n]$ into a discrete-time output signal $y[n]$.

$$x[n] \xmapsto{\mathcal{H}} y[n] \quad \text{or} \quad y[n] = \mathcal{H}\{x[n]\},$$

Classes of systems



The three classes of system: analog systems, digital systems, and interface systems from analog-to-digital and digital-to-analog.

Interface systems

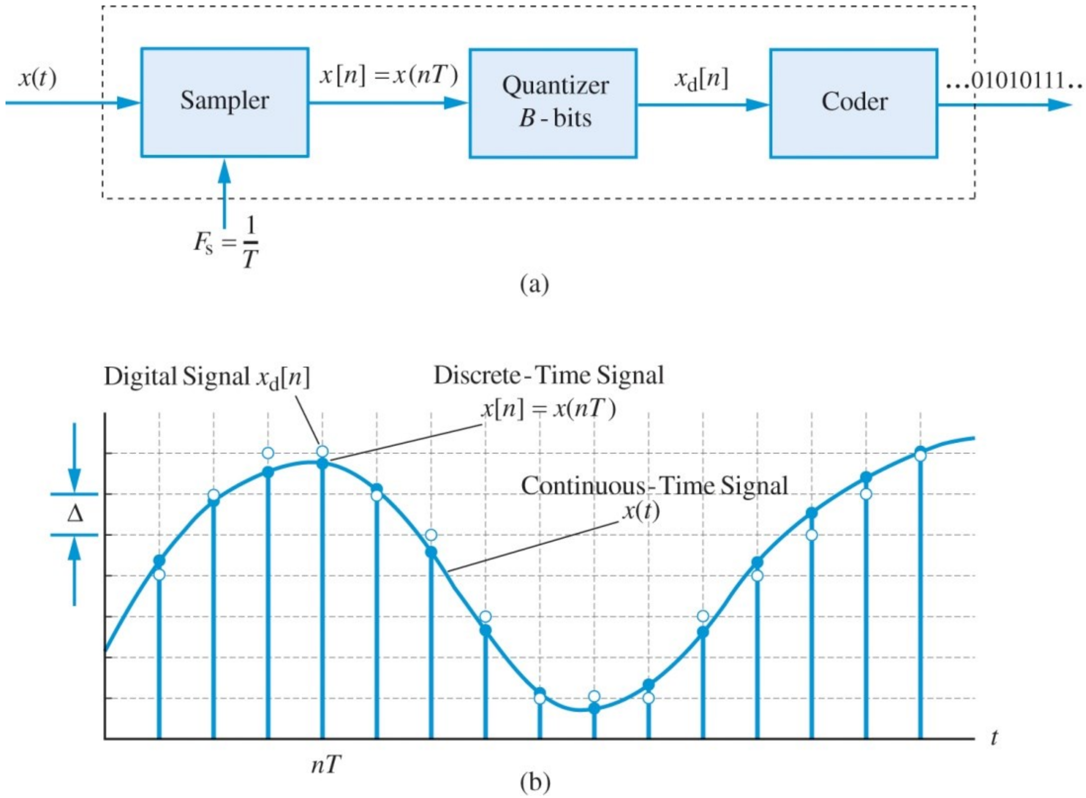
Analog-to-digital conversion

Conceptually, the conversion of an analog (continuous time, continuous-amplitude) signal into a digital (discrete-time, discrete-amplitude) signal, is a simple process; it consists of two parts:

Sampling converts a continuous-time signal to a discrete-time signal by measuring the signal value at regular intervals of time.

Quantization converts a continuous-amplitude x into a discrete-amplitude x_d. The result is a digital signal that is different from the discrete-time signal by the quantization error or noise.

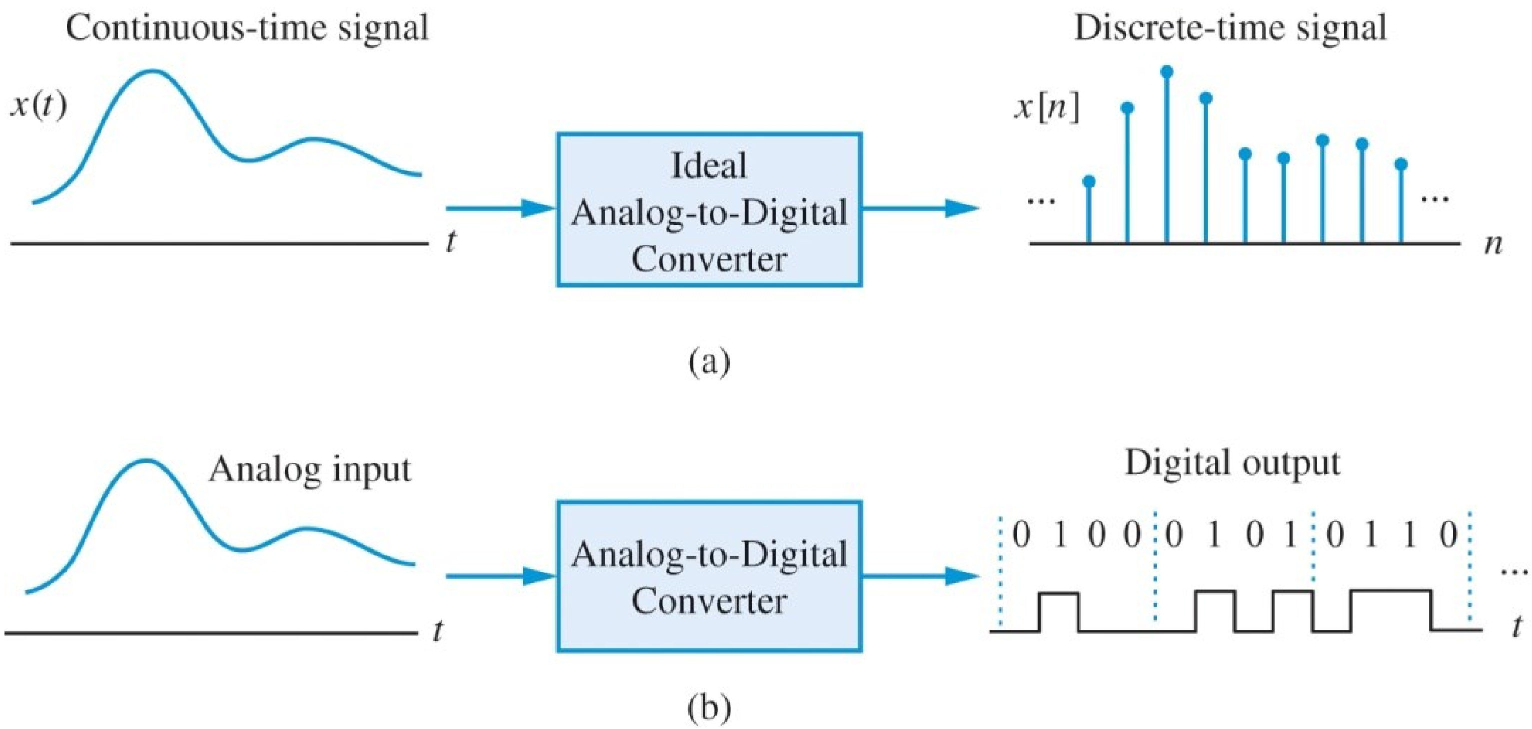
Analog-to-digital conversion



(a) Block diagram representation of the analog-to-digital conversion process. (b) Examples of the signals *x(t)*, *x*[*n*], and *x*d[*n*] involved in the process. The amplitude of *x*[*n*] is known with infinite precision, whereas the amplitude of *x*d[*n*] is known with finite precision (quantization step or resolution).

Interface systems

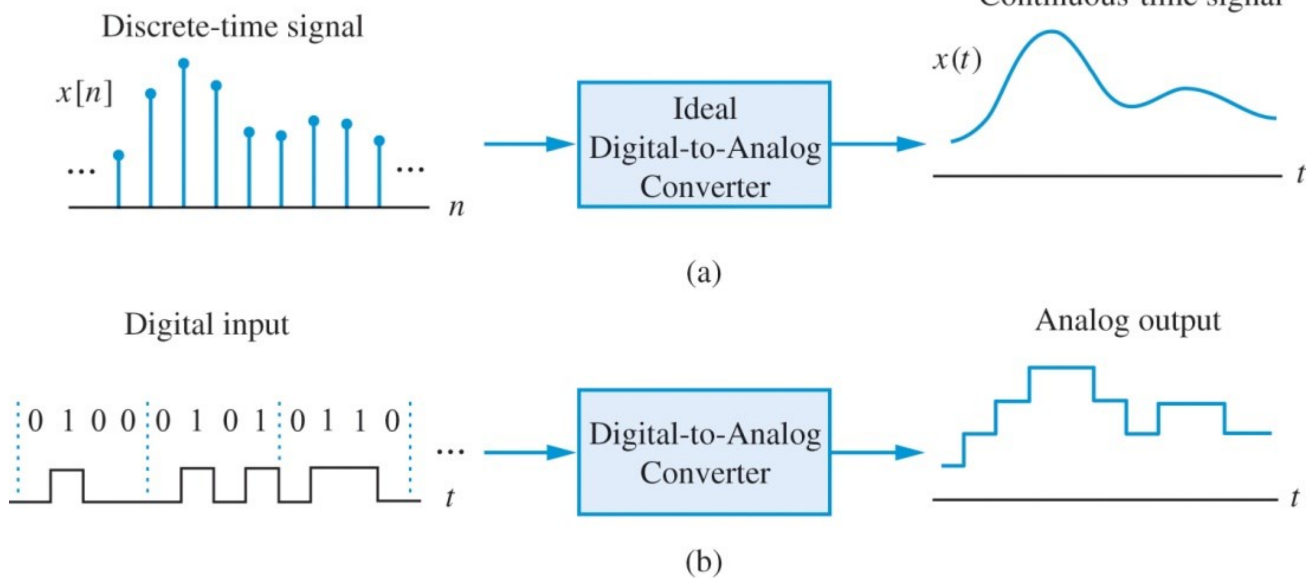
Analog-to-digital conversion



Block diagram representation of an ideal (a) and a practical (b) analog-to-digital converter, and the corresponding input and output signals. The input to the ideal ADC is a function and the output is a sequence of numbers; the input to the practical ADC is an analog signal and the output is a sequence of binary code words. The number of bits B, in each word, determines the accuracy of the converter.

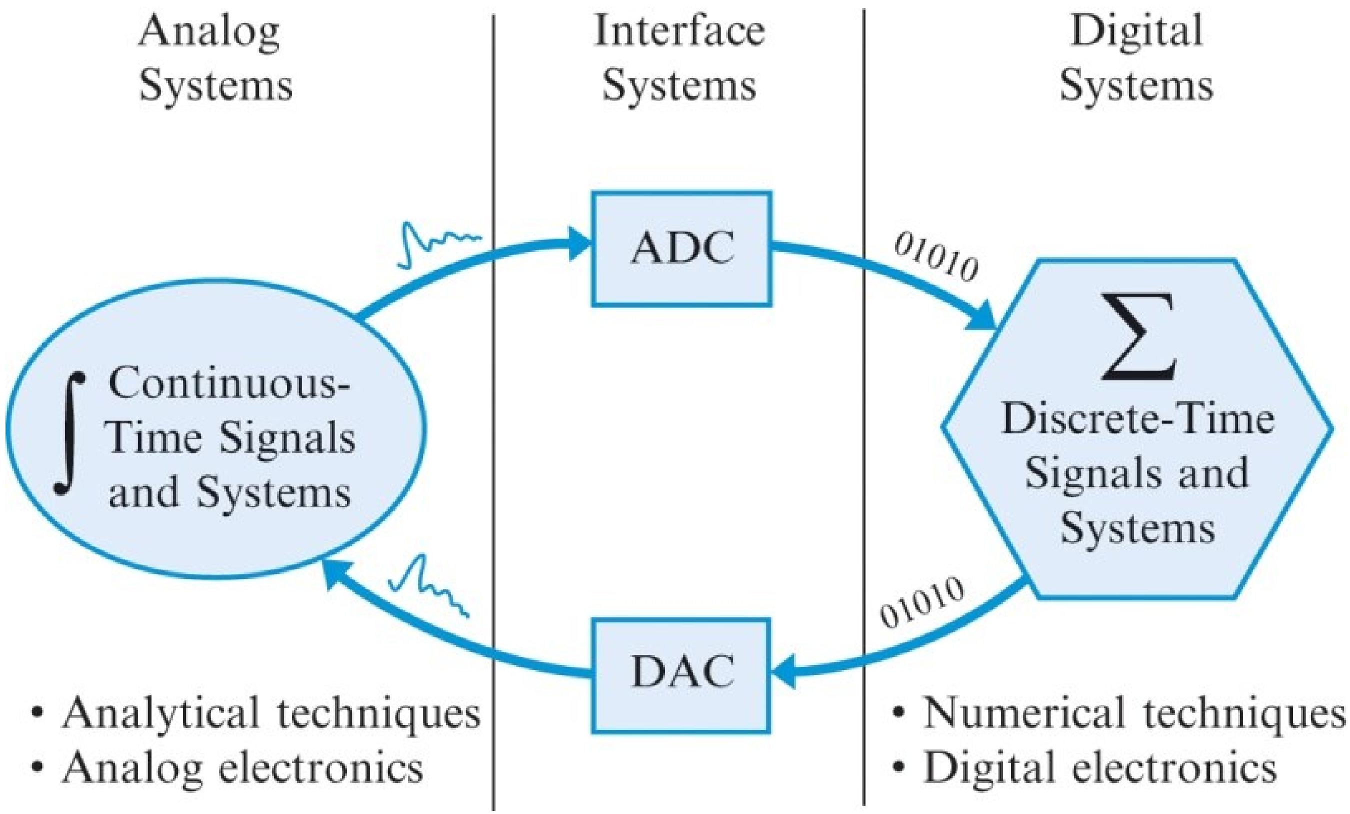
Digital-to-analog conversion

The *ideal D/A converter* or *interpolator* is essentially filling the gaps between the samples of a sequence of numbers to create a continuous-time function. A practical DAC takes a value represented in digital code and converts it to a voltage or current that is proportional to the digital value.



Block diagram representation of an ideal D/A converter (a) and a practical D/A converter (b) with the corresponding input and output signals. In most practical applications, the staircase output of the D/A converter is subsequently smoothed using an analog reconstruction filter.

Classes of systems



The three classes of system: analog systems, digital systems, and interface systems from analog-to-digital and digital-to-analog.

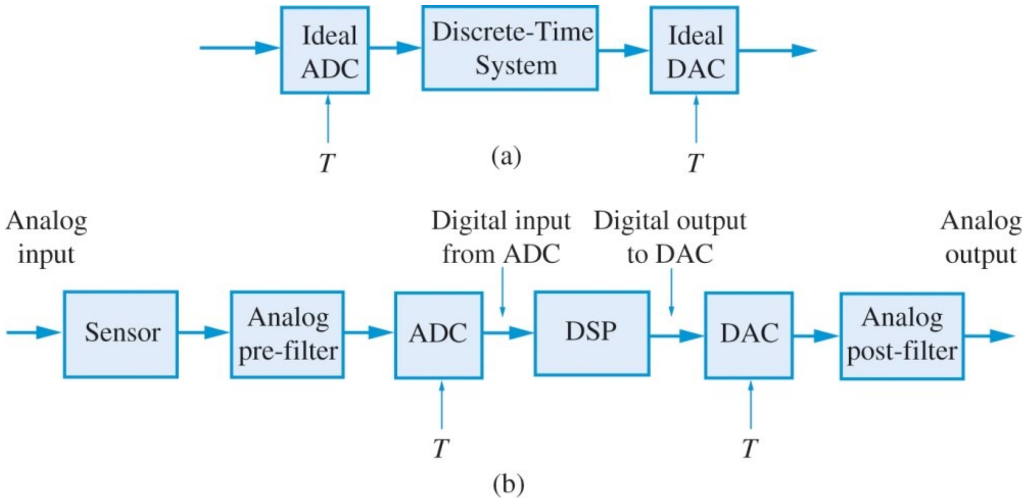
Digital signal processing

In theory, where we concentrate on the essential mathematical aspects of signal processing, we deal with ideal (infinite precision) discrete-time signal processing systems, and ideal A/D and D/A converters.

In practice, due to inherent real-world limitations, a typical system for the digital processing of analog signals includes the following parts

1. A sensor that converts the physical quantity to an electrical variable. The output of the sensor is subject to some form of conditioning, usually **amplification**, so that the voltage of the signal is within the voltage sensitivity range of the converter.
2. An analog filter (known as pre-filter or antialiasing filter) used to “**smooth**” the input signal before sampling to avoid a serious sampling artifact known as **aliasing** distortion

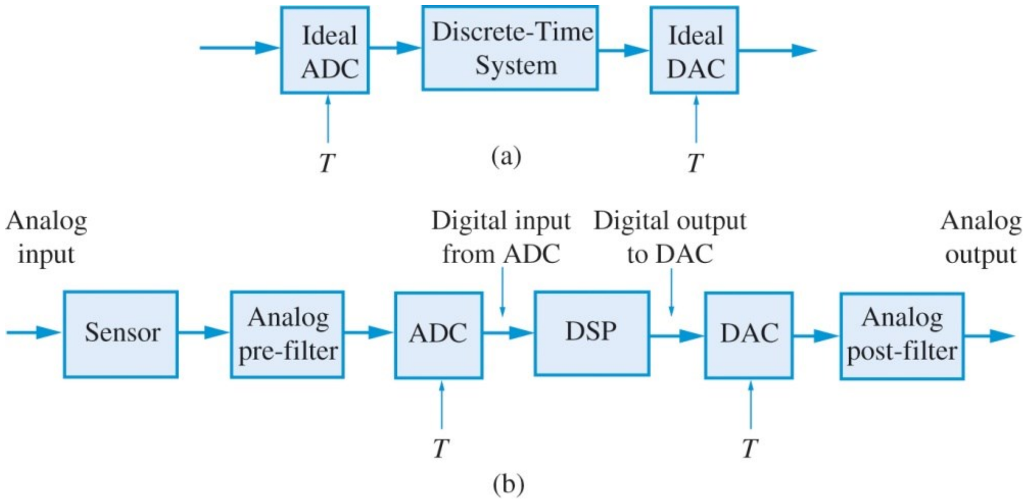
Simplified block diagram of idealized system for (a) continuous-time processing of discrete-time signals, and (b) its practical counterpart for digital processing of analog signals.



Digital signal processing

3. An A/D converter that converts the analog signal to a digital signal. After the samples of a discrete-time signal have been stored in memory, time-scale information is lost. The **sampling rate** and the **number of bits** used by the ADC determine the accuracy of the system.
4. A digital signal processor (DSP) that executes the signal processing algorithms. The DSP is a **computer chip** that is similar in many ways to the microprocessor used in personal computers. A DSP is, however, designed to perform certain numerical computations extremely **fast**. Discrete-time systems can be implemented in real-time or off-line, but ADC and DAC always operate in real-time. *Real-time* means completing the processing within the allowable or available time between samples.

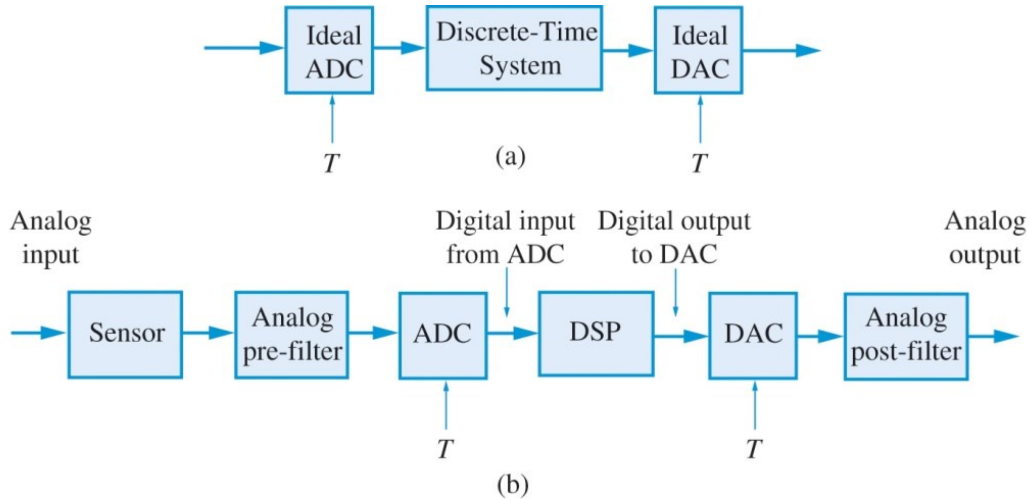
Simplified block diagram of idealized system for (a) continuous-time processing of discrete-time signals, and (b) its practical counterpart for digital processing of analog signals.



Digital signal processing

5. A D/A converter that converts the digital signal to an analog signal. The DAC, which reintroduces the lost time-scale information, is usually followed by a sample-and-hold circuit. Usually, the A/D and D/A converters operate at the **same sampling rate**.
6. An analog filter (known as **reconstruction** or **anti-imaging filter**) used to smooth the staircase output of the DAC to provide a more faithful analog reproduction of the digital signal

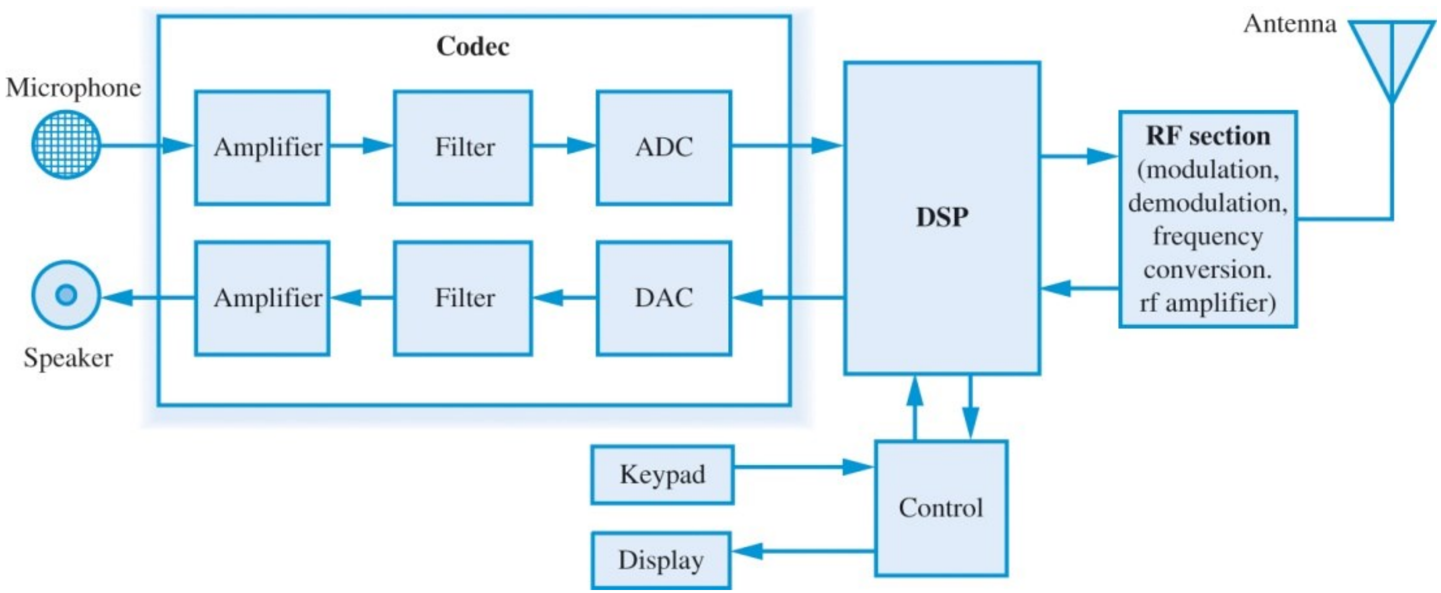
Simplified block diagram of idealized system for (a) continuous-time processing of discrete-time signals, and (b) its practical counterpart for digital processing of analog signals.



Advantages of Digital signal processing

1. Sophisticated signal processing functions can be implemented in a cost-effective way using digital techniques.
2. There exist important signal processing techniques that are difficult or impossible to implement using analog electronics.
3. Digital systems are inherently more reliable, more compact, and less sensitive to environmental conditions and component aging than analog systems.
4. The digital approach allows the possibility of time-sharing a single processing unit among a number of different signal processing functions.

A typical application



Simplified block diagram of a digital cellular phone.

Typical signal processing operations

Generally, the desired operations on signals are implemented by a combination of some elementary operations:

Scaling

the multiplication of a signal either by a positive or negative constant.

In the case of analog signals, the operation is usually called **amplification** if the magnitude of the multiplying constant, called **gain**, is greater than 1.

If the magnitude of the multiplying constant is less than 1, the operation is called **attenuation**.

Delay

generates a signal that is a delayed replica of the original signal.

For an analog signal $x(t)$, $y(t) = x(t-t_0)$ is the signal obtained by delaying $x(t)$ by the amount t_0 of time which is assumed to be a positive number.

If t_0 is negative, then it is an **advance** operation.

Typical signal processing operations

The **integration** of an analog signal x(t) generates a signal

$$y(t) = \int_{-\infty}^{t} x(\tau)d\tau$$

The **differentiation** of an analog signal x(t) generates a signal

$$w(t) = \frac{dx(t)}{dt}$$

Typical signal processing operations

More complex operations are implemented by combining two or more elementary operations.

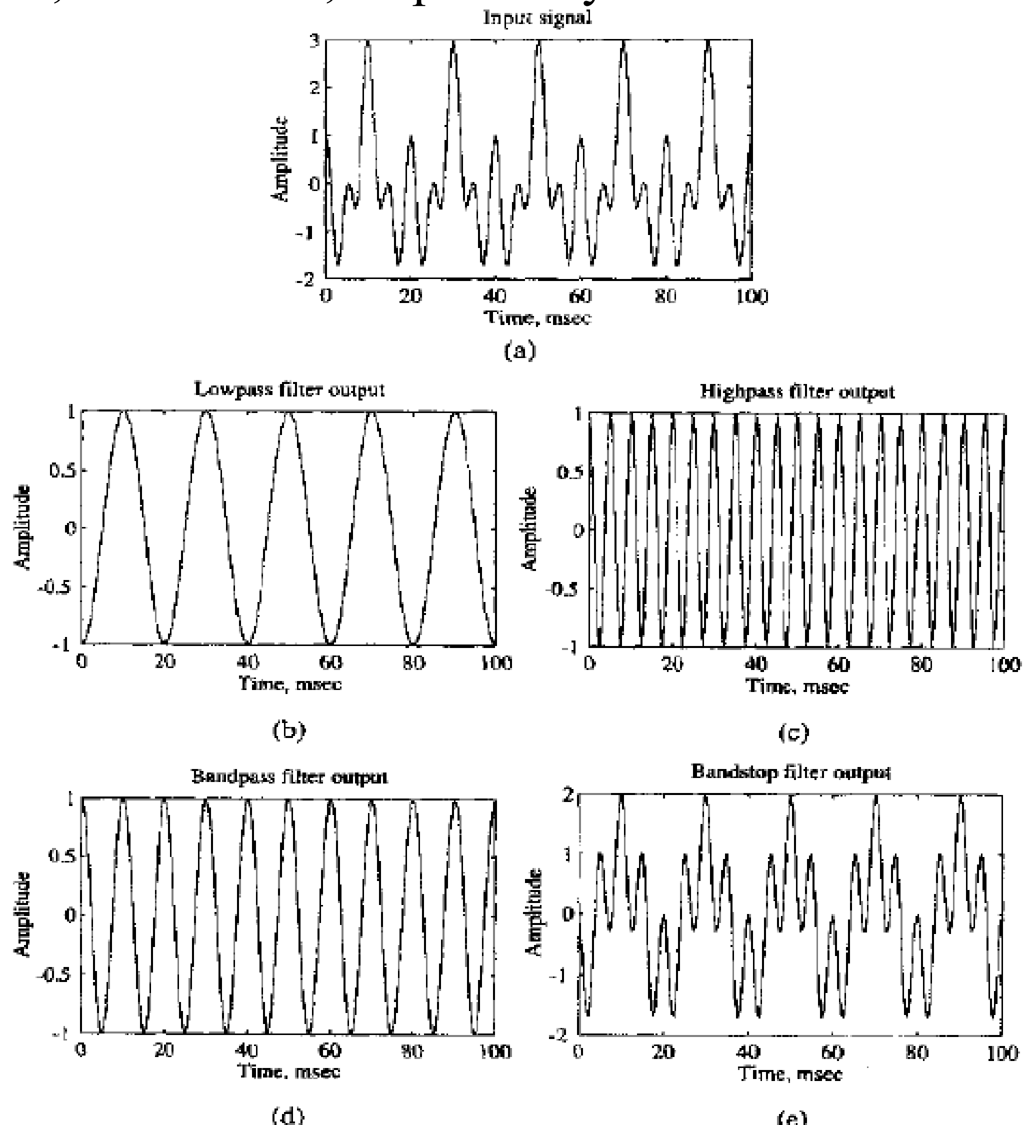
Filtering

Filtering is one of the most widely used complex signal processing operations. A filter passes certain frequency components without any distortion and blocks other frequency components.

The range of frequencies that is allowed to pass through the filter is called the passband, and the range of frequencies that is blocked by the filter is called the stopband.

Typical signal processing operations

A signal composed of three sinusoidal components of frequencies 50 Hz, 110 Hz, and 210 Hz, respectively.

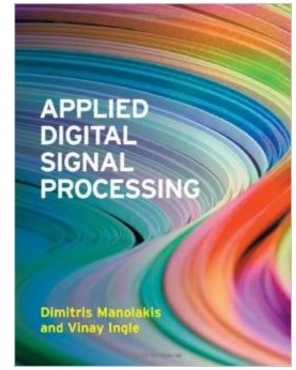


Syllabus

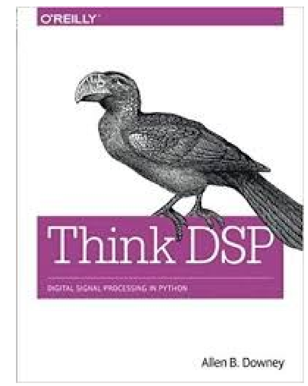
- Introduction
- Discrete–time signals and systems
- The z-transform
- The discrete-time Fourier transform
- Sampling of continuous-time signals
- Design of FIR digital filters
- Design of IIR digital filters
- DCT, PCA, Wavelet transforms

Textbooks

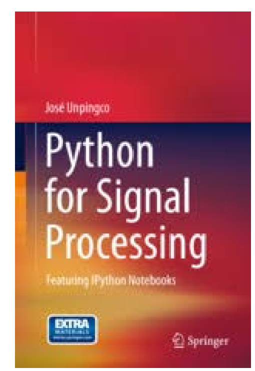
Manolakis, Dimitris G., and Vinay K. Ingle. ***Applied digital signal processing: theory and practice***. Cambridge University Press, 2011.



Downey, Allen B. ***Think DSP: digital signal processing in Python***. O'Reilly Media, Inc.



Unpingco, José. ***Python for Signal Processing***. Springer International Pu, 2016.springer



Programming tool

Python

- www.python.org
- Scientific Python distributions

  Enthought Canopy

  Anaconda

  Python(x,y)

  WinPython

  Pyzo

  Algorete Loopy

Learning material

1. "Python for Signal Processing" By José Unpingco. Available for students through UC Berkeley Library Here
2. Python Bootcamp by Josh Bloom and Fernando Perez Here
3. Introduction to Python (general) Here
4. A Crash Course in Python for Scientists Here
5. Scientific Computing with Python Here
6. Using Python for Signal Processing and Visualization Here

Grading

Homeworks:	15%
Project:	20%
Midterm:	30%
Final exam:	35%