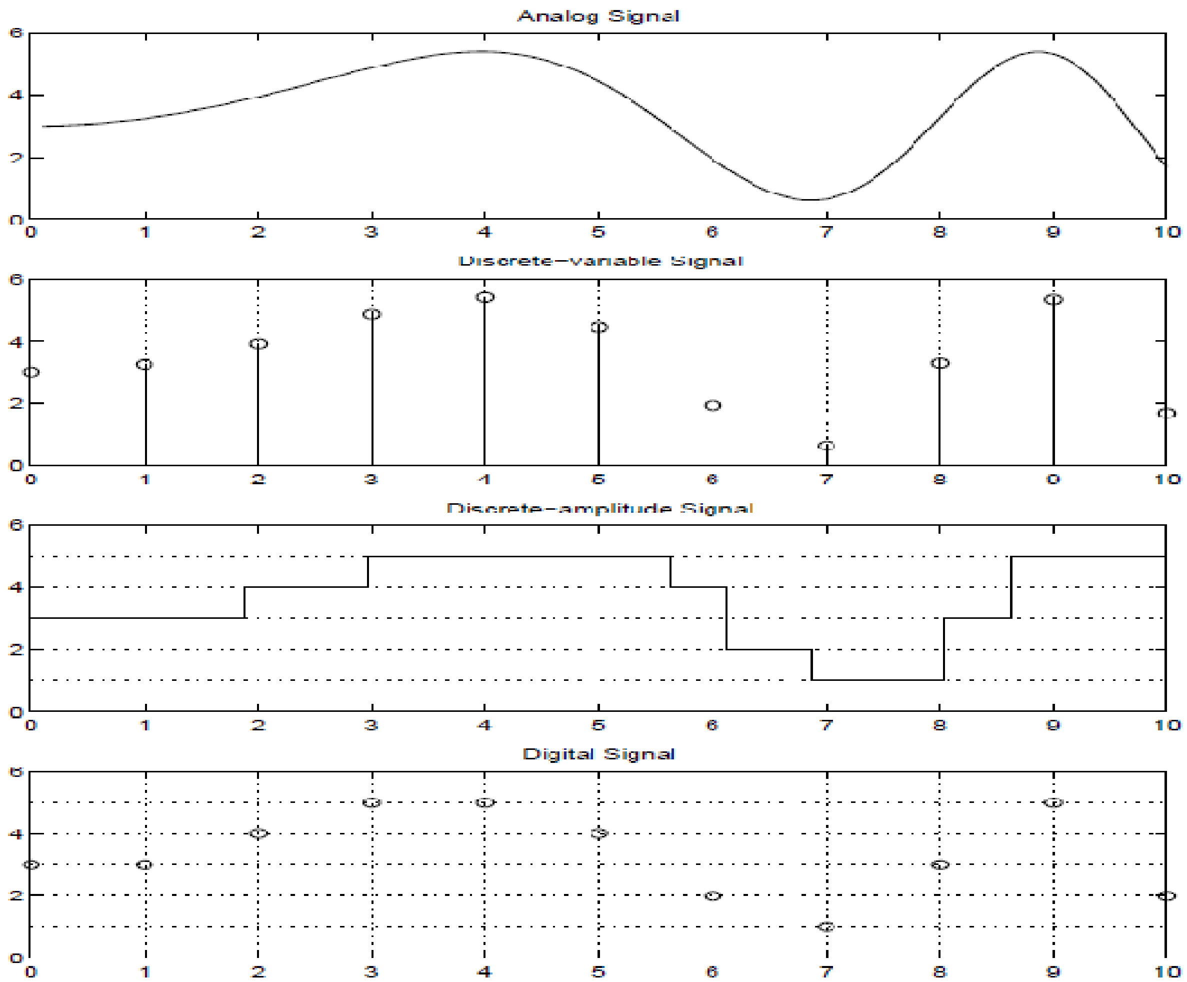
SAMPLING AND DSP - AN INTRODUCTION

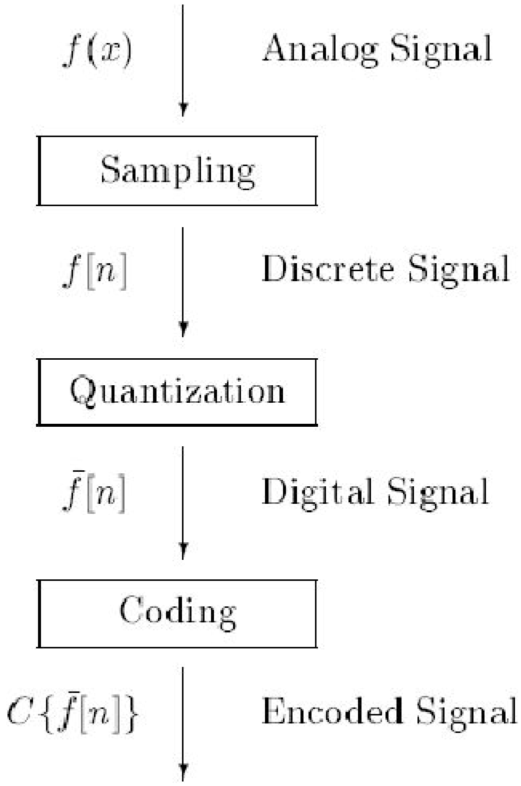
SAMPLING:

The signals we use in the real world, such as our voices, are called "analog" signals. To process these signals in computers, we need to convert the signals to "digital" form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. To convert a signal from continuous time to discrete time, a process called sampling is used. The value of the signal is measured at certain intervals in time. Each measurement is referred to as a sample. (The analog signal is also quantized in amplitude; This is known as the Nyquist rate. The Sampling Theorem states that a signal can be exactly reproduced if it is sampled at a frequency F, where F is greater than twice the maximum What happens if we sample the signal at a frequency that is lower that the Nyquist rate? When the signal is converted back into a continuous time signal, it will exhibit a phenomenon called *aliasing*. Aliasing is the presence of unwanted components in the reconstructed signal.



Basic Ideas in Sampling Theory

● *sampling* a signal: Analog → Digital conversion by reading the value at discrete points



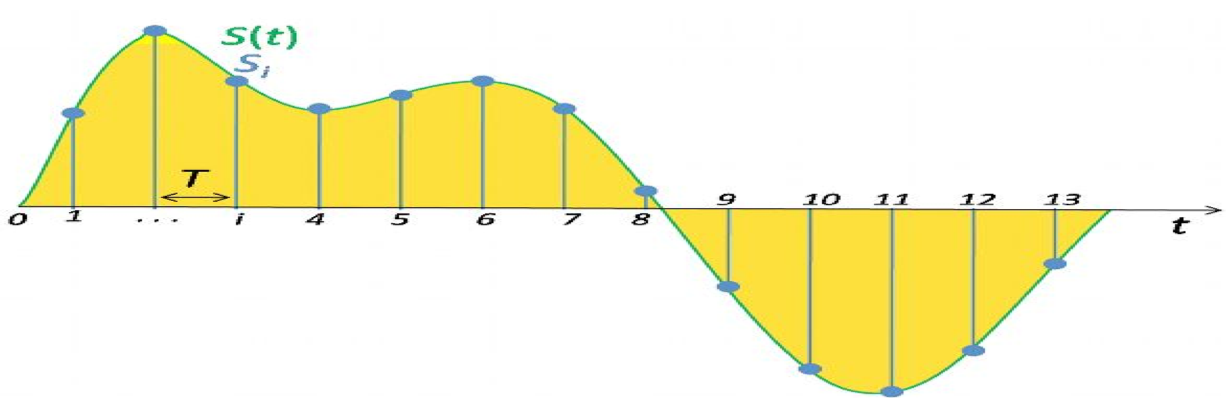
Digital signal processing (DSP):

In its most general form, DSP refers to the processing of analog signals by means of discrete-time operations implemented on digital hardware.

• From a system viewpoint, DSP is concerned with mixed systems:

- the input and output signals are analog

- the processing is done on the equivalent digital signals.



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