Colorado State University

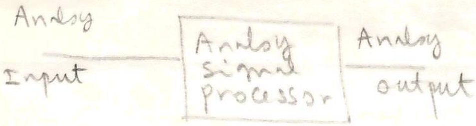
Course No. EE 512

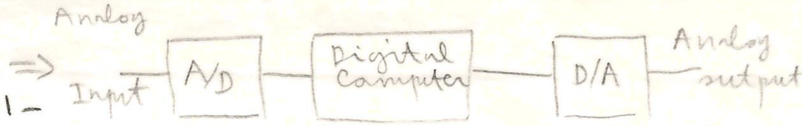
Lecture No. 1

Date Sept. 1, 1992

Instructor M.R. AZIMI

Assignment Start Reading CH. 1 and 2 Sections 1.1–1.3 and 2.1–2.5.





can it be done in real-time?

Digital Signal Processing (EE 651)

Signal processing, in general has rich history and its importance is evident in such diverse fields as biomedical engineering, acoustics, Sonar, radar, speech communications, data Compression, nuclear science and many others.

Until recently signal processing has typically been carried out using analog equipments. Later the use of digital computers arose in different ways. Before implementing in analog hardware digital computer can be used to simulate a signal processing system. This simulation offers tremendous advantages in flexibility. However, the processing could not be done in real-time. Example of this type of simulation is the early work on digital filtering. The filtering operation is programmed on a digital computer so that an analog signal can pass through an A/D converter, be filtered and then be transformed back to analog using a D/A Converter. There was a natural tendency to experiment with increasingly sophisticated signal processing algorithms. Some of these grew out of the flexibility of the digital Computer and had no apparent practical implementation in analog equipments.

The evolution of a new point of view toward DSP was further accelerated by the disclosure (1965) of an efficient algorithm for Computation of Discrete Fourier Transform. This algorithm has Come to be known as the FFT. Apart from the significant

Applications requiring very large wide bandwidths require analog/optical signal proc.

Saving that FFT offers, it is inherently a discrete-time process i.e. it is not simply an approximation to a continuous time Fourier Transform. Instead it gives set of properties and mathematics for discrete-time signals. This indeed represents a shift away from the notion that signal processing on digital computer was merely an approximation to analog signal processing. Thus, DSP area was born with the emergence of FFT technique and is growing without limit with advances in VLSI technologies.

The improvements in VLSI technology in recent years has had an important impact on the field of DSP. Many sophisticated algorithms can be implemented on special purpose hardwares which can handle sampling rates in MHZ range. FFT algorithms can be implemented on special purpose hardware on a real-time rate.

The importance of DSP appears to be increasing with no visible sign of saturation. Indeed, the future developments are promising to be more dramatic than the present ones. While DSP is a dynamic, rapidly growing field, its fundamentals are well formulated. Many of these stemmed from the classical control and sampled data control systems in 1940's and 50's and also numerical analysis techniques developed in the 60's.

In this course we will cover the fundamental techniques and methods that are technology independent to provide a firm understanding of this field in order to appreciate for its wide scope of applications and future directions.

Terminology

A signal is a function which conveys information, generally about the state or behaviour of a physical system. Signals can be categorized into four groups

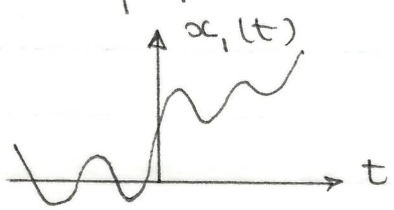
1) Continuous - time 2) Discrete - time signals

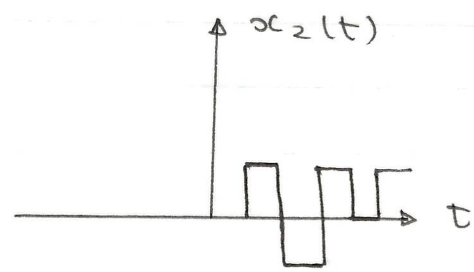
3) Analog signals 4) Digital signals

These are defined in terms of an independent variable, $x(t)$ which is time in temporal signal processing and distance (space) in spatial signal processing applications.

$x(u,v)$

1- **Continuous - time Signals**: function in which information is described over a continuous range of time. The amplitude either ranges over a continuous range of values or a finite number of possible values.



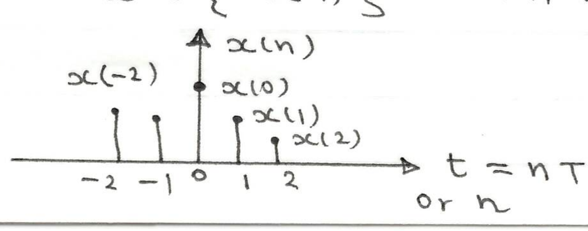


2- **Discrete - time Signals**: function in which information is described over a certain set of values of time

$$x = \{x(n)\} \qquad |n| < \infty$$

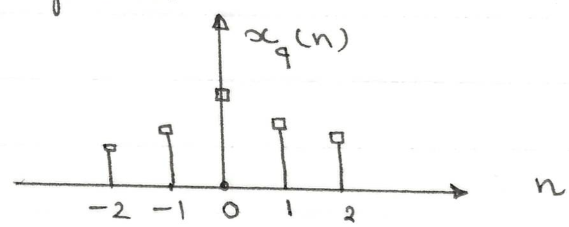
$x(n)$: nth sample

$\{x(n)\}$: the sequence $x(n)$



3 - Analog Signal : Continuous - time signal whose amplitude can take a continuous range of values. (Continuous - valued)
Term analog introduced in connection with analog computers, where voltages and currents are used to represent physical quantities e.g. velocity, displacement, etc. $x_1(t)$ is analog but $x_2(t)$ is not.

4 - Digital Signal : discrete - time signal whose amplitude can take only a certain set of values i.e. both time and amplitude are discrete (sampled and quantized).
Quantization level is determined by the type and word - length of the computer system used.



Systems

A system is composed of several basic components which operate together to perform a desired task.

1 - Continuous - time systems : A system in which the input and output signals are both continuous - time.

2 - Discrete - time system : A system in which the input and output signals are both discrete - time.

3 - Analog System : A system in which the input and output signals are both analog.

4- Digital System : A system in which both input and output signals are digital

Advantages of Sampling in Systems

Some of the major advantages of sampling in systems are

tolerance

1) improved sensitivity performance 2) better reliability

3) no drift 4) Less effect due to noise and disturbance

5) more compact and lightweight 6) less cost

7) more flexibility in programming

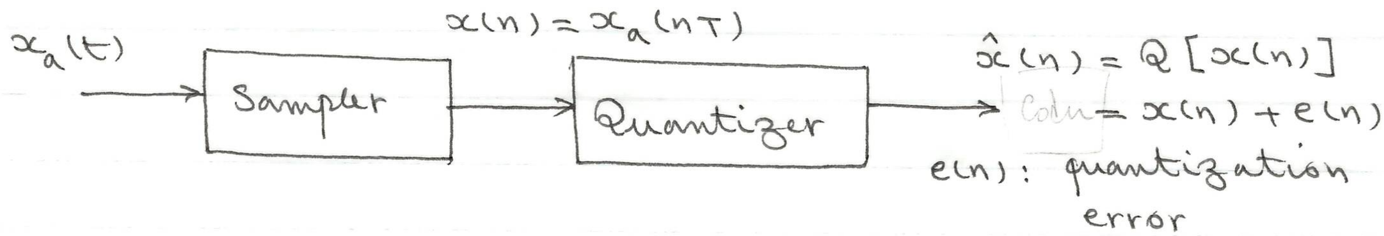
Easy to store and retrieve in Digital form.

Digital filters are more versatile than the analog ones. The program which characterizes a digital filter can be modified to accommodate design changes, or adaptive performances without variations on the hardware. Digital components

such as registers, multipliers, etc are often more reliable and more rugged in construction and more compact in size than their analog counterparts. The disadvantage of digital systems are inaccuracies due to quantization effects namely roundoff error and truncation.

Mapping Between Analog and Digital Domains

The process of mapping an analog signal to a digital signal involves two separate processes that are sampling and quantization.



The sampler takes a bandlimited analog signal $x_a(t)$ and converts it to a discrete-time signal in which each sample is represented with infinite precision i.e. with infinite number of bits. Of course, physical limitations of computers preclude sampling with infinite precision, thus each samples must be truncated or rounded to fit a finite-length register. This action is performed by the quantizer which generates, from $x(n)$, a quantized $\hat{x}(n)$. We have

$$\hat{x}(n) = x(n) + e(n)$$

where $e(n)$ represents the quantization effects. Since almost all the mathematical tools required for the analysis of digital systems applies equally to discrete-time systems we do not consider the effects of quantizations until towards the end of the term. Thus we consider discrete-time signals that are generally represented by

$$\{x(n)\}_{n \in R} = \text{Sequence of } x(n)$$

R : Set of possible values for n (integers)