Signal Processing for AI

Course Objectives

- 본 교과목은 샘플링 이론을 포함한 신호처리 기본 원리와 이론, 아날로그 및 디지털 신호, 그리고 다양한 변환의 기본 개념을 다룬다. 신호의 시간 및 주파수 공간의 특성을 푸리에 변환과 z 변환 방법을 도입하여 해석한다. 유한 및 무한 임펄스 응답 필터를 정의하고 그 특성을 선형 시불변 시스템에 대해 해석한다. 또한, 확률 및 랜덤 프로세스 등의 정의와 이를 다루기 위한 확률밀도함수, 평균, 상관함수, 전력 스펙트럼 및 이의 통계신호처리 응용 예에 대해 다룬다.

Some useful information (1)

- Instructor
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    - hpark@sogang.ac.kr
  - Phone
    - 02-705-8916
  - Office hours
    - Mon. & Fri. 10:00~12:00, Wed. 15:00~17:00

- TA **이승현**
  - E-mail
    - ee_seung@u.sogang.ac.kr
  - Phone
    - 02-711-8916

Some useful information (2)

- Homepage
  - http://eclass.sogang.ac.kr
  - Lab
    - http://iip.sogang.ac.kr
- Textbook
  - Signal Processing First
    - Authors: J. H. McClellan, R. W. Schafer, and M. A. Yoder
    - Publisher: Pearson Education, Inc.
- Reference
  - B. Porat, “A Course in Digital Signal Processing,” John Wiley & Sons, Inc.
  - O. C. Ibe, “Fundamentals of Applied Probability and Random Processes,” Elsevier.

Evaluation

- Mid-term exam (50%), final exam (50%)
- You should read the original(English) version of the textbook.
- You should read the textbook for the coming lecture in advance. It is very important for you to understand the lecture.
- Problem-solving homeworks will be given at the end of almost every lectures.
- You will be quizzed in the beginning of almost every lectures.

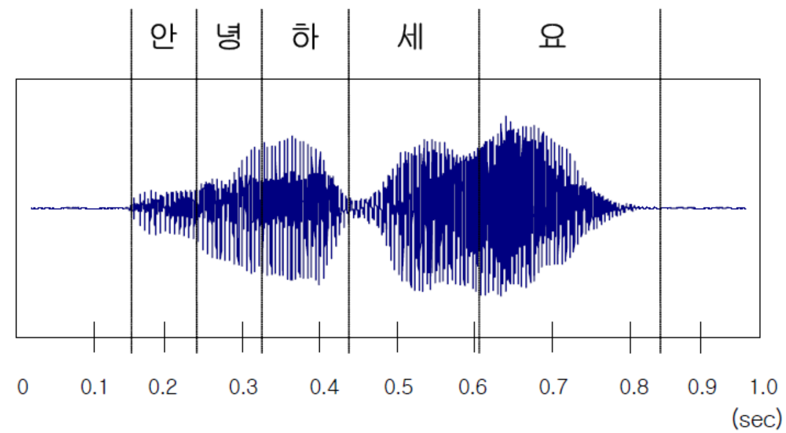
Others

- Any kinds of protests or appeals for scoring or grading (including FA) will not be accepted if they are caused by your individual problem.
- You have to be here before class. The FA rule will be applied very strictly.
- All submitting materials should be submitted by the due date.
- You must turn off your cell phone when you are in the class. You must not use any electronic devices without permission.
- You are required to meet me at least one time until the end of this semester.

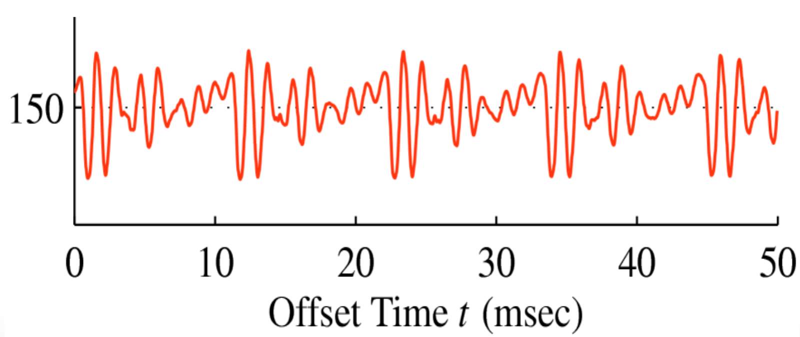
Chapter 1
Introduction

Mathematical Representation of Signals

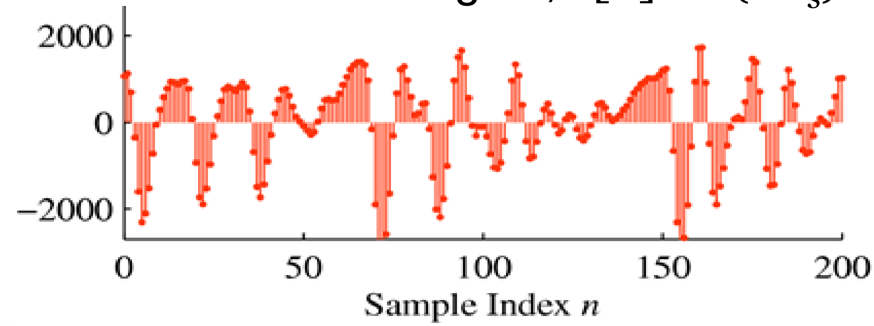
- Signals
  - Patterns of variations that represent or encode information.
- Speech signal



Continuous-time signal, $s(t)$

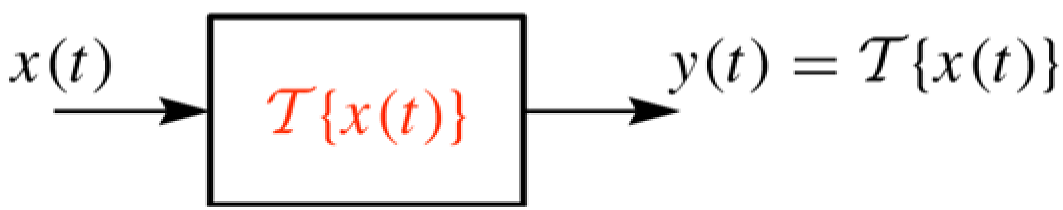


Discrete-time signal, $s[n] = s(nT_s)$



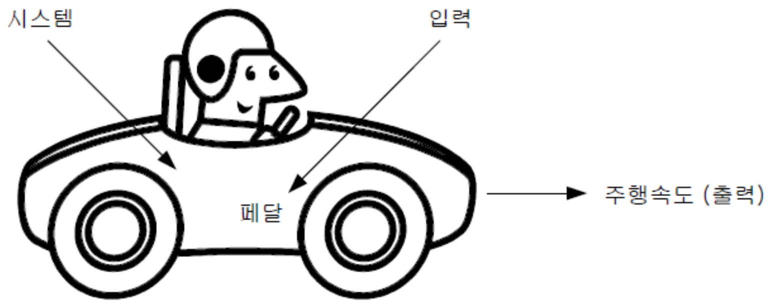
Mathematical Representation of Systems

- A system
  - Something that transforms signals into new signals or different signal representations.
- One-dimensional continuous-time system
  - $y(t) = \mathcal{T}\{x(t)\}$
  - $y(t)$: output signal, $x(t)$: input signal
  - The input signal is operated on by the system (symbolized by the operator $\mathcal{T}$) to produce the output.
  - Block diagram representation of the system

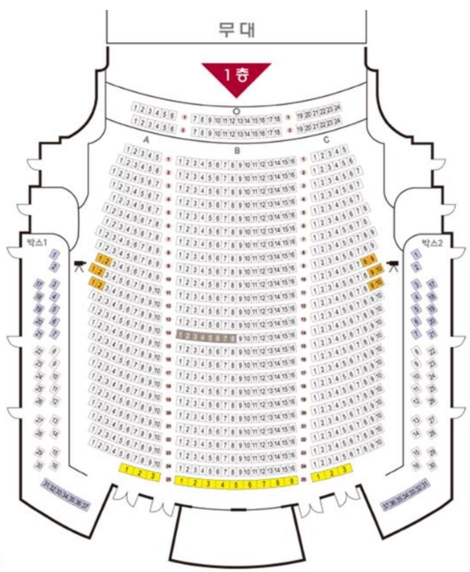


Examples of Systems

- Car

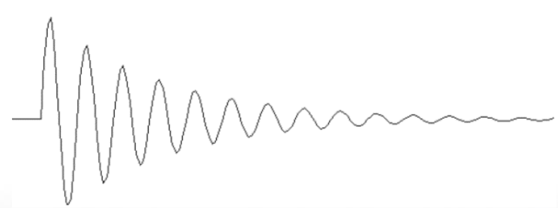


- Opera House



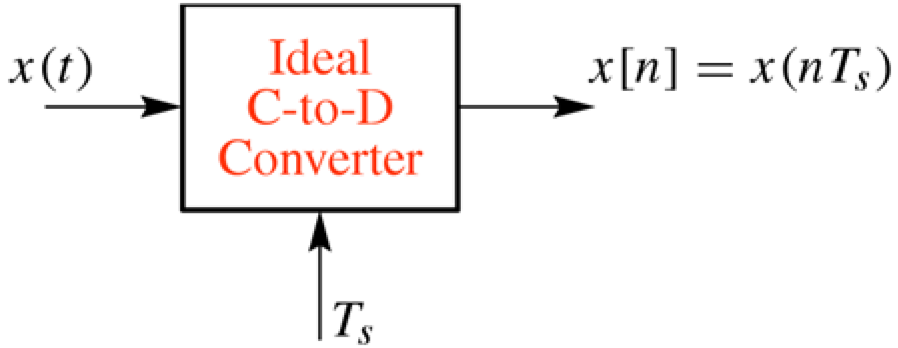
Input signal

output signal

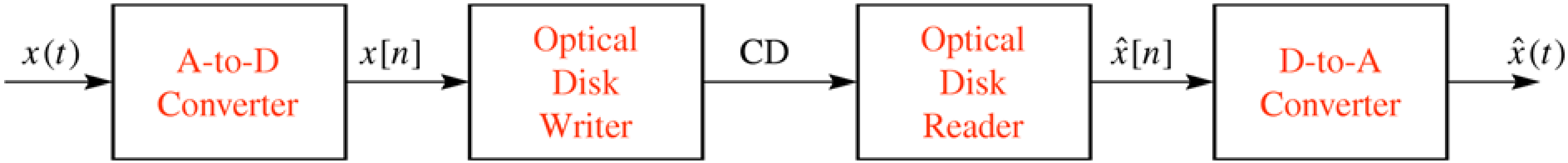


Other Examples of Systems

- Sampler
  - Convert a continuous-time signal into a discrete-time signal

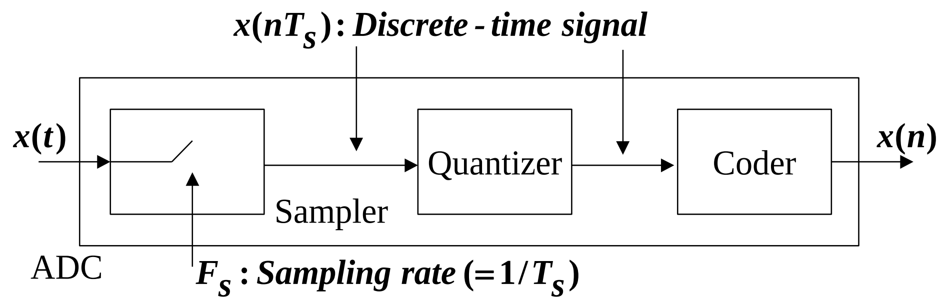


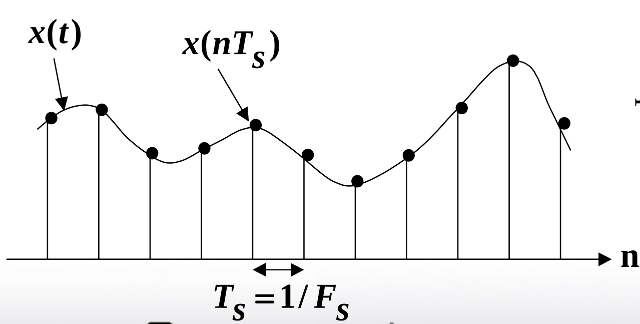
- Block diagram for recording and playback of an audio CD

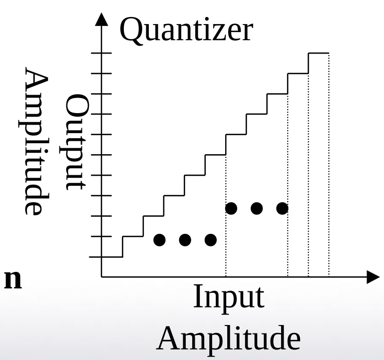


A/D(Analog-to-Digital) Conversion

- Digital signal
  - Sampling
    - Convert a continuous-time signal into a discrete-time signal
  - Quantization
    - Discretize the amplitudes of samples for the discrete-time signal
  - Represented by a sequence of numbers or symbols

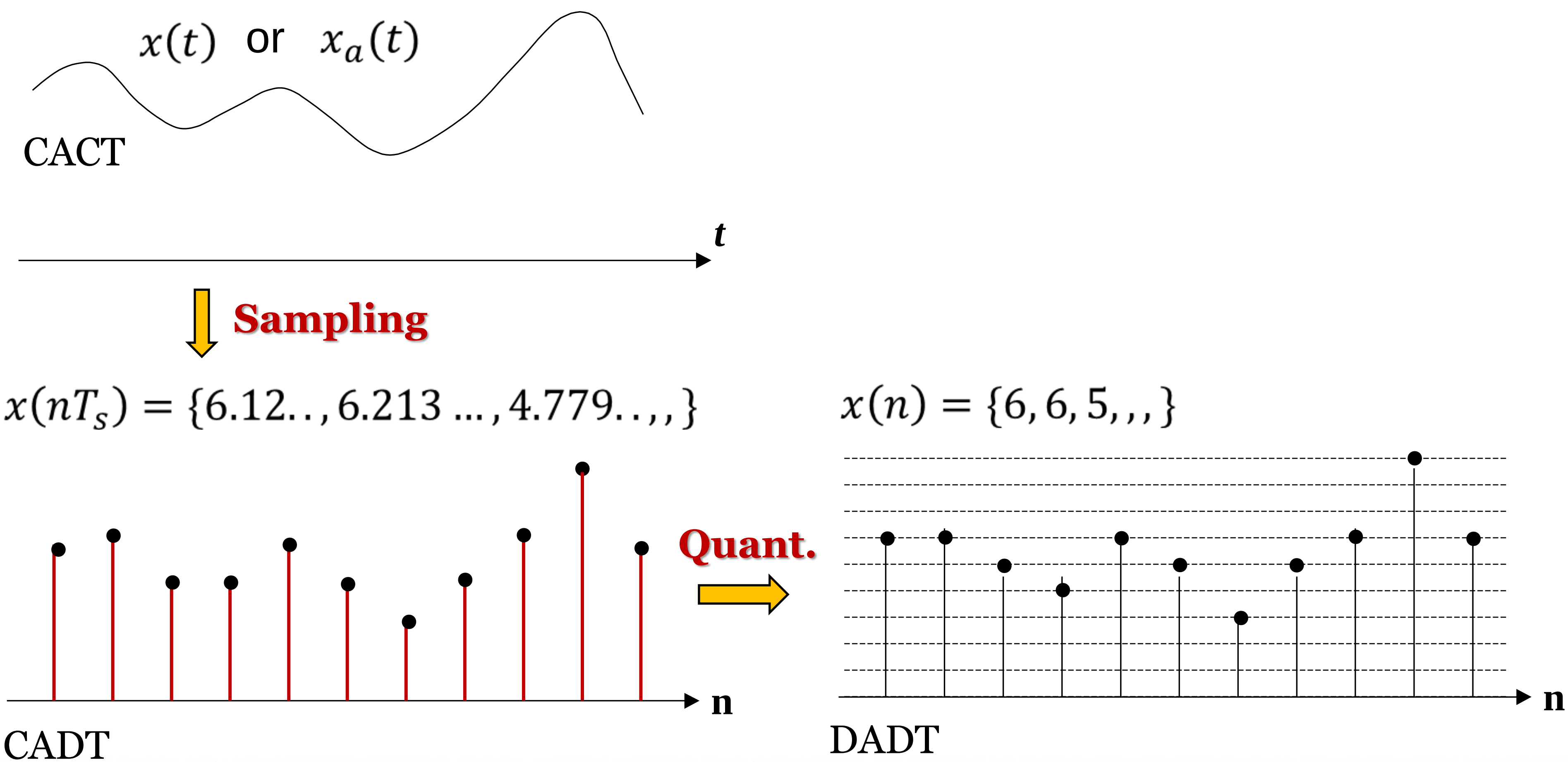






A/D(Analog-to-Digital) Conversion

- Sampling and quantization



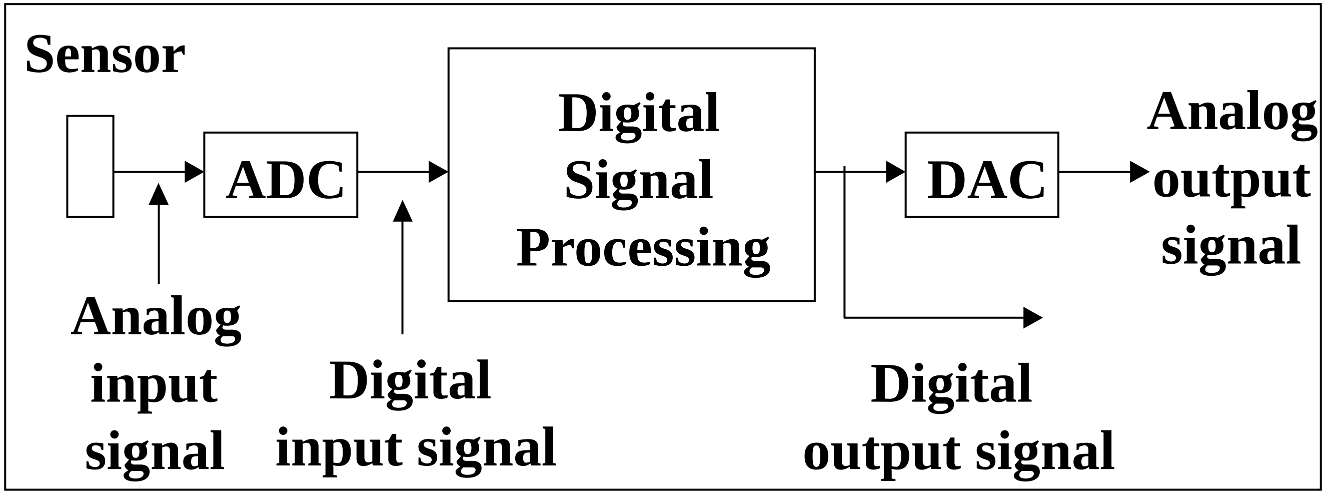
Digital Signal Processing

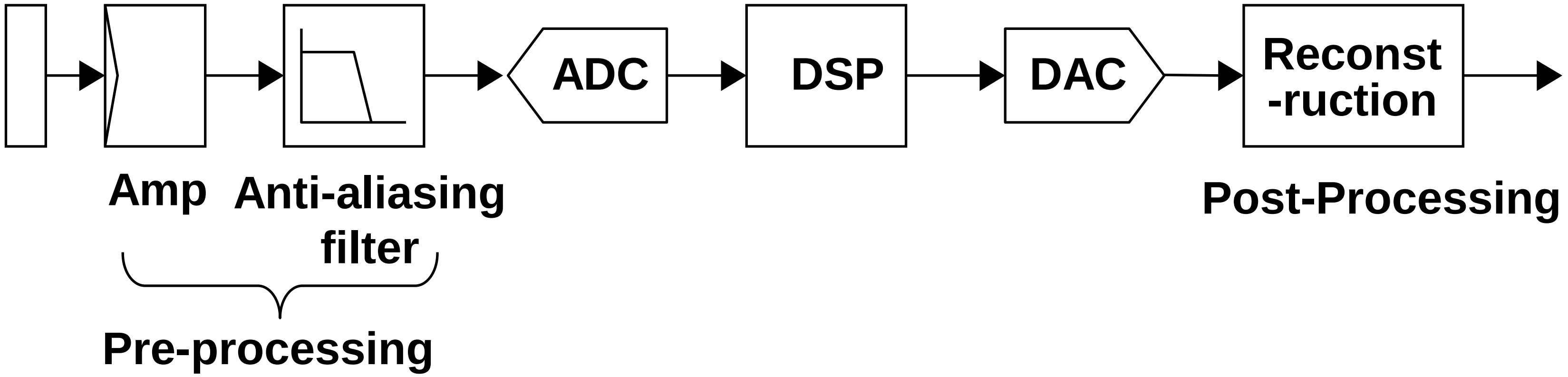
- DSP (digital signal processing)
  - Processing of the digital signals
  - Examples
    - Audio and speech signal processing, sonar and radar signal processing, sensor array processing, digital image processing, signal processing for communications, control of systems, biomedical signal processing, seismic data processing, etc
- Why DSP?
  - Computer processing
  - New signal type that computers can handle

Processing of Real-World Signals Using DSP

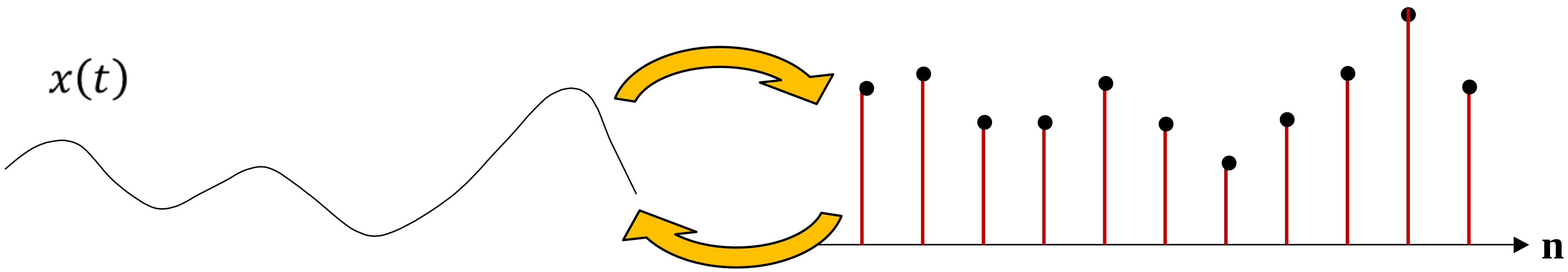
- Real-world signals
  - Continuous analog signals
- Processing of the real-world signals using DSP
  - First step
    - Convert the analog signal into a digital form, by sampling and quantizing it using an analog-to-digital converter (ADC), which turns the analog signal into a stream of numbers
  - Second step
    - Transform a stream of numbers into another stream of numbers by DSP
  - Final step
    - Convert the transformed stream of numbers into an analog output signal, which requires a digital-to-analog converter (DAC)

DSP System


Sensor
ADC
Digital
Signal
Processing
DAC
Analog
output
signal
Analog
input
signal
Digital
input signal
Digital
output signal


ADC
DSP
DAC
Reconst
-ruction
Amp
Anti-aliasing
filter
Pre-processing
Post-Processing

Reconstruction



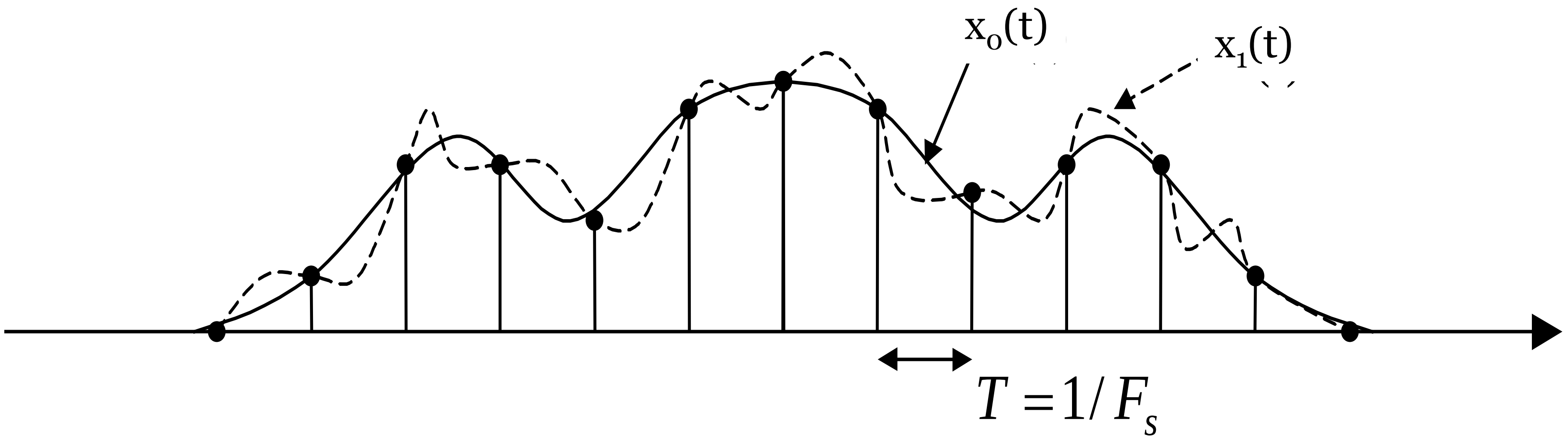
A/D conversion or sampling rule ?

➔ What happens when sampling a signal ?

How to reconstruct the analog signal from the sampled data ?

➔ Can we perfectly reconstruct the original analog input signal?

Reconstruction

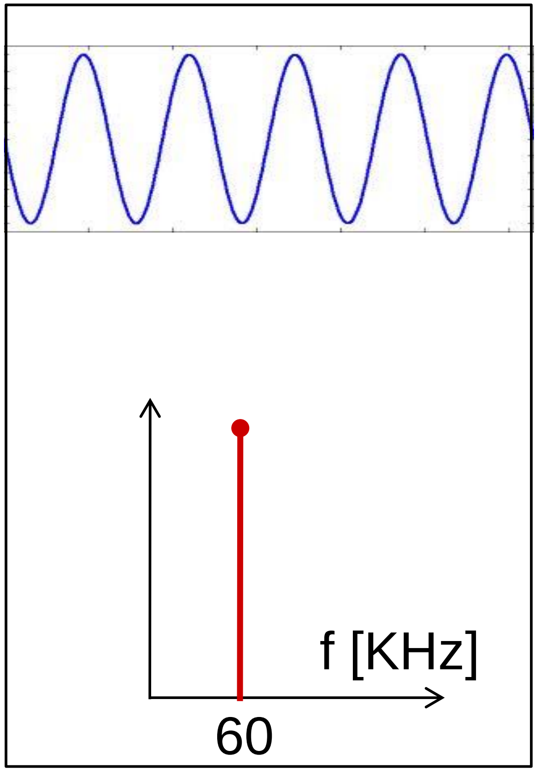


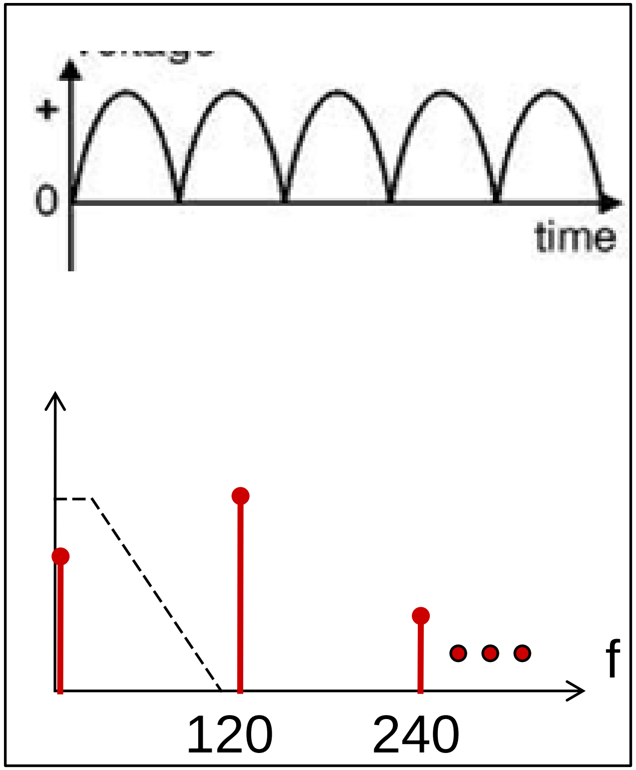
1. The following two CT signals $x_0(t)$ and $x_1(t)$ are different, but their samples are identical.
2. Then, what does the sample sequence represent?
   What will be reconstructed from the sampled data?

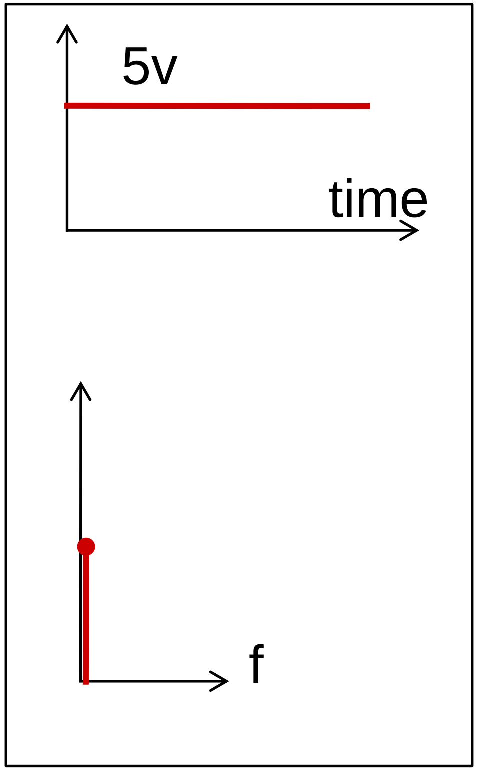
You will learn the theories related to this question and their applications

Frequency Domain Analysis

- Frequency components of the following waveforms?







Fourier Series and Transform

- Fourier series
  - For a periodic signal

$$x(t) = \sum_{k=-\infty}^{\infty} a_k e^{j2\pi k f_0 t}$$

- Fourier transform
  - For a non-periodic signal

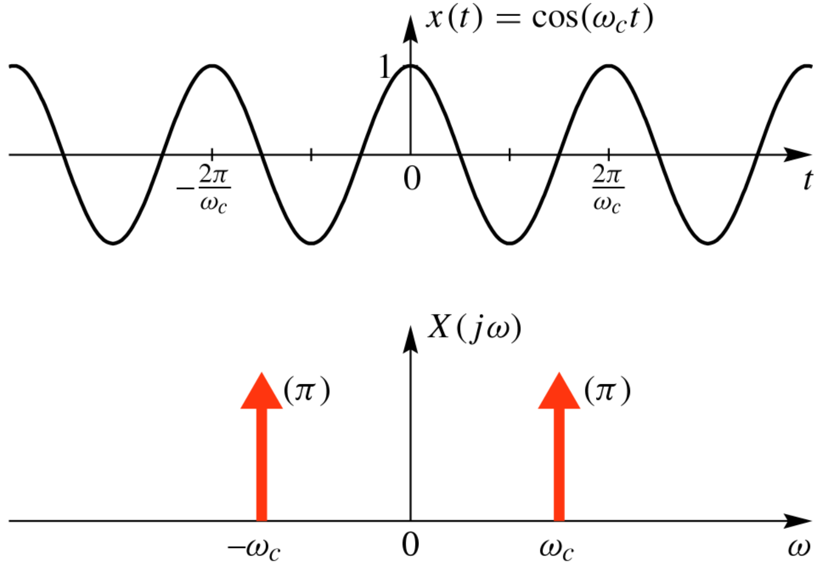
$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(j\omega) e^{j\omega t} d\omega \qquad X(j\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$

$$\text{Time - Domain} \Leftrightarrow \text{Frequency - Domain}$$

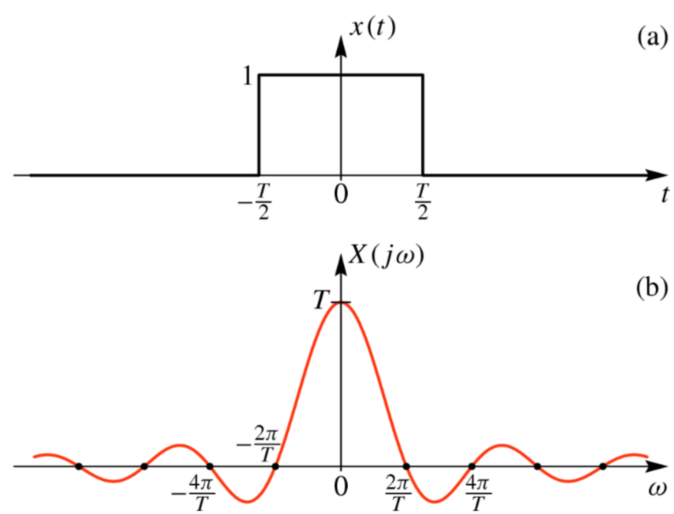
$$x(t) \Leftrightarrow X(j\omega)$$

Example of Transforms

- Sinusoid



- Rectangular pulse



Frequency domain analysis: analog vs. digital

Analog signal vs. Digital signal

(CTCA, CTDA, DTCA, DTDA)

Continuous-time signals : periodic and aperiodic

Discrete-time signals : periodic and aperiodic

Laplace transform	⟷	Z-transform
Fourier Transform (FT)	⟷	Discrete-Time Fourier Transform (DTFT)
Fourier Series (FS)	⟷	Discrete Fourier Transform (DFT) / FFT

Thank you

- Reading assignment
  - ~ Section 2-4