SIO 207A: Fundamentals of Digital Signal Processing
Class 1

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0

Digital Signal Processing

- **Signal Processing Applications:** Entertainment, acoustics, geophysics, space exploration, communications, medicine, and archaeology
- Signal processing algorithms and hardware are prevalent in a wide range of systems from low-cost high-volume consumer electronics to highly specialized military systems
- A goal of this class was to enable graduate students at SIO that have never taken a fundamentals of DSP class (e.g., ECE161A) to take more advanced graduate-level DSP classes (SIO 207B, SIO207C, SIO207D)
- A detailed syllabus is available online

1

1

Overview

- **Online Platform:** Handouts and homework assignments will be posted on the *Canvas calendar*
- **Class Schedule:** Lectures are Mondays and Wednesdays 3:30PM – 4:50PM in SPIESS 330. The class on Wednesdays is followed by a discussion session from 5:00PM – 5:50PM.
- **Office Hours:** Office hours are every *Friday at 3 PM* via Zoom (a link to the Zoom meeting will be posted on the Canvas calendar)
- **Grades:** Homework 30%, midterm project 30%, and final project 40%

2

2

Homework and Research Projects

- **Homework assignments and projects** to be solved in MATLAB will be posted approximately every 1-2 weeks on Canvas and will be due one week later
- The first three homework problems can be solved by using basic MATLAB functionalities
- Collaborations are encouraged but the developed MATLAB solutions you hand in should reflect your own understanding of the course material
- The assigned **mid-term and final project** should be considered as take-home exams; they should represent individual effort and assistance should not be given nor received from anyone other than the instructor

3

3

Guidelines for Homework and Research Projects

- Style of a technical report or mini conference paper
- Use text editor (Word, Tex, …) to write the report
- ``Tell story'' by using plots and add some text to complement the story
- Label all plots, add Figure caption, and mention them in the text
- Report should be well-organized and self-contained
- Plot should always have labels on both axis
- Note that auto-scaling option of MATLAB can give unsatisfying results

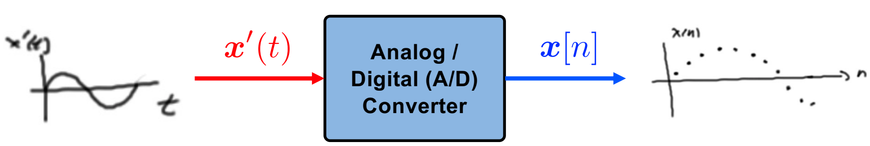
Bibliography

- *Discrete-Time Signal Processing*, Alan V. Oppenheim and Ronald W. Schafer, Prentice Hall, 2009.

Slides based on lecture notes develop by Prof. William S. Hodgkiss and additional references will be posted on the Canvas website of the course

Questions?

Analog / Digital Conversion



samples & quantizes

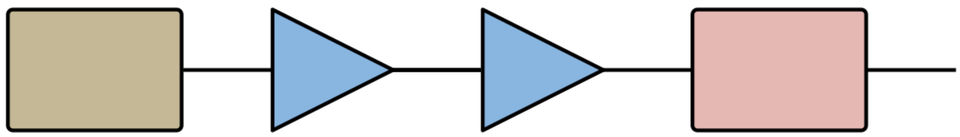
$$x[n] = x'(nT) = \sum_{k=-\infty}^{\infty} x'(kT)\,\delta(n-k)$$

"Unit sample sequence"

sampling rate: f_s
sampling interval: $T = 1/f_s$

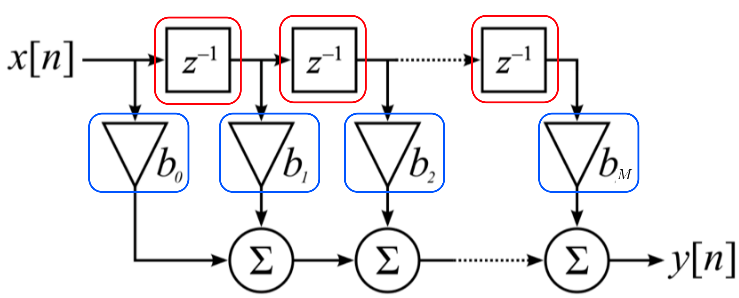
Unit sample (Impulse):
$\delta(n-k) = 1$ for $n = k$
$\delta(n-k) = 0$ otherwise

Signal Conditioning Before A/D Conversion



Filtering of Time-Series

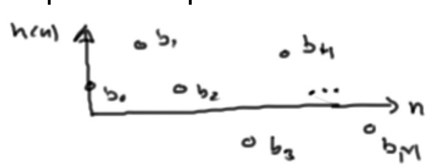
- Finite Impulse Response (FIR) Digital Filters



Unit Delay

Coefficients $b_k, k \in \{0, 1, \ldots, M\}$

Impulse Response:



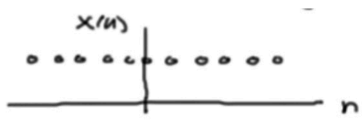
Filtering of Time-Series

- Output of FIR Filter:

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_M x[x-M]$$
$$= \sum_{k=0}^{M} b_k x[n-k]$$

Coefficients b_k also referred to as "weights"

- Consider the following input signals to the filter



``ultimate" low-frequency signal

``ultimate" high-frequency signal (alternating sequence)

10

10

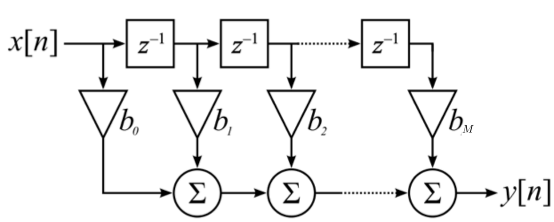
Some Basic FIR Filters

- **A:** $b_0 = 1, b_k = 0$ for $k > 0$
  all-pass filter

- **B:** $b_k = 1$ for $k = k'$ and $b_k = 0$ otherwise
  pure delay of k' units (unit magnitude response and linear phase response)

- **C:** $b_k = 1/(M+1), k \in \{0, \ldots, M\}$
  low pass characteristic

- **D:** $b_k = 1/(M+1), \quad k$ even
  $b_k = -1/(M+1), \quad k$ odd
  high pass characteristic



11

11

FIR Filters

- **A: Unit sample response, i.e.,** $h[n] = b_n$
  (analogous to impulse response)
- **B: Generates only zeros**
  Example: low pass filter with all $b_k = 1$
  $x[n]$ is an unit alternating sequence
  Number of zeros: M

Convolution:

$$y[n] = h[0]x[n] + h[1]x[n-1] + \dots h[M]x[n-M]$$
$$= \sum_{k=0}^{M} h[k]x[n-k]$$

- **C: Filter description (input/output relationship)**

Difference Equations:

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_M x[n-M]$$
$$= \sum_{k=0}^{M} b_k x[n-k]$$

12

12