EE327 Digital Signal Processing Overview

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Why are you here?

1. Learn DSP
2. Pass exams
3. Have some time to other subjects
4. Have fun!!

Your final goal ……

Find the optimal balance between these four CONFLICTING goals

Teaching Team

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Course Syllabus

- Introduction
- Overview & application of DSP
- Discrete – time signals & systems
- The discrete – Fourier transform.
- The FFT algorithm
- The Z- transform
- Digital filters
- Design of FIR & IIR filters
- Realization of digital filters.

You need forever to learn all of these!!!!!!!!!!!!!!!!!!

Where does it fit in your field

- DSP is a core course for communications engineer.
- DSP is an essential course for a computer engineer.
- DSP is an important practical course for a power engineer

Course Philosophy

- Maximize practical engineering sense
- Maximize field exposure
- Minimize complex mathematics

You need to USE DSP effectively not to INVENT new DSP algorithms or concepts

Text Books

Main Text

- The scientist and engineer's guide to Digital Signal Processing
  - 2nd edition by Steven Smith (Sm)

- Digital Signal Processing – Filtering Approach
  - 1st edition by Steve White (Wh)

Other References

- Discrete Time Signal Processing
  - 2nd edition by Oppenheim and Schafer (Op)

Course Content (tentative)

1.	What is DSP (Introduction)	(Wh CH1,Sm CH2)	2/20
2.	Statistics, Probability and Noise	(Sm CH 2)	2/27
3.	ADC and DAC	(Sm CH 3)	3/06
4.	Linear Systems	(Sm CH 5)	3/13
5.	Convolution	(Sm CH 6, 7)	3/20
6.	Discrete Fourier Transform	(Sm CH 8, 9)	3/27
7.	Fourier Transform Properties &FFT	(Sm CH 10,12)	4/03
8.	Z Transform	(Wh CH 4,5)	4/10
9.	Digital Filtering	(Sm CH 14)	4/17
10.	Moving Average Filter (FIR)	(Sm CH 15)	4/24
11.	Windowed Sync Filters	(Sm CH 16)	5/01
12.	Recursive Filters (IIR)	(Sm CH 19)	5/08

Subject to modification any time during the semester

Prerequisites

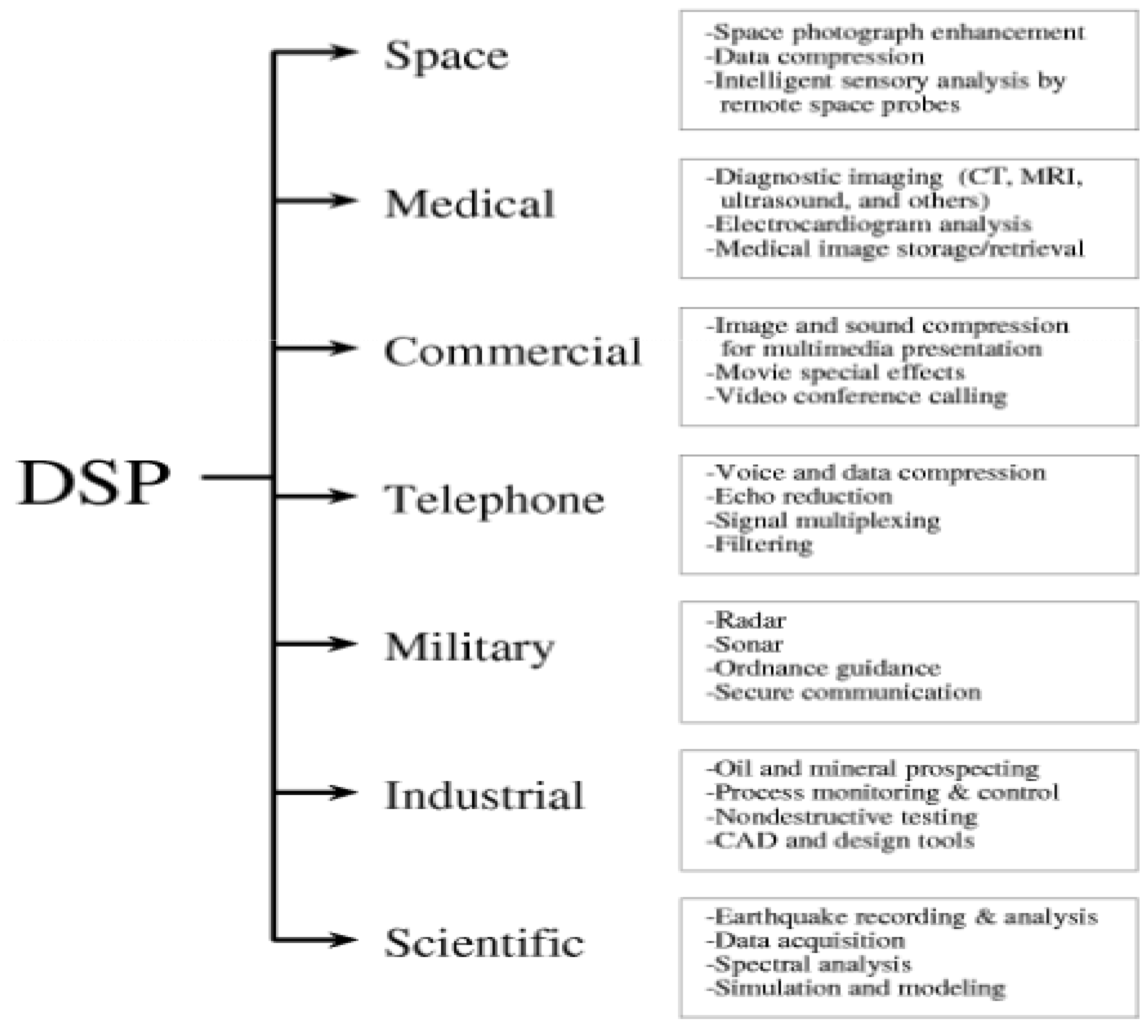
- A course in computer programming
- A course in basic electronics
- A course in calculus

- The mindset to learn

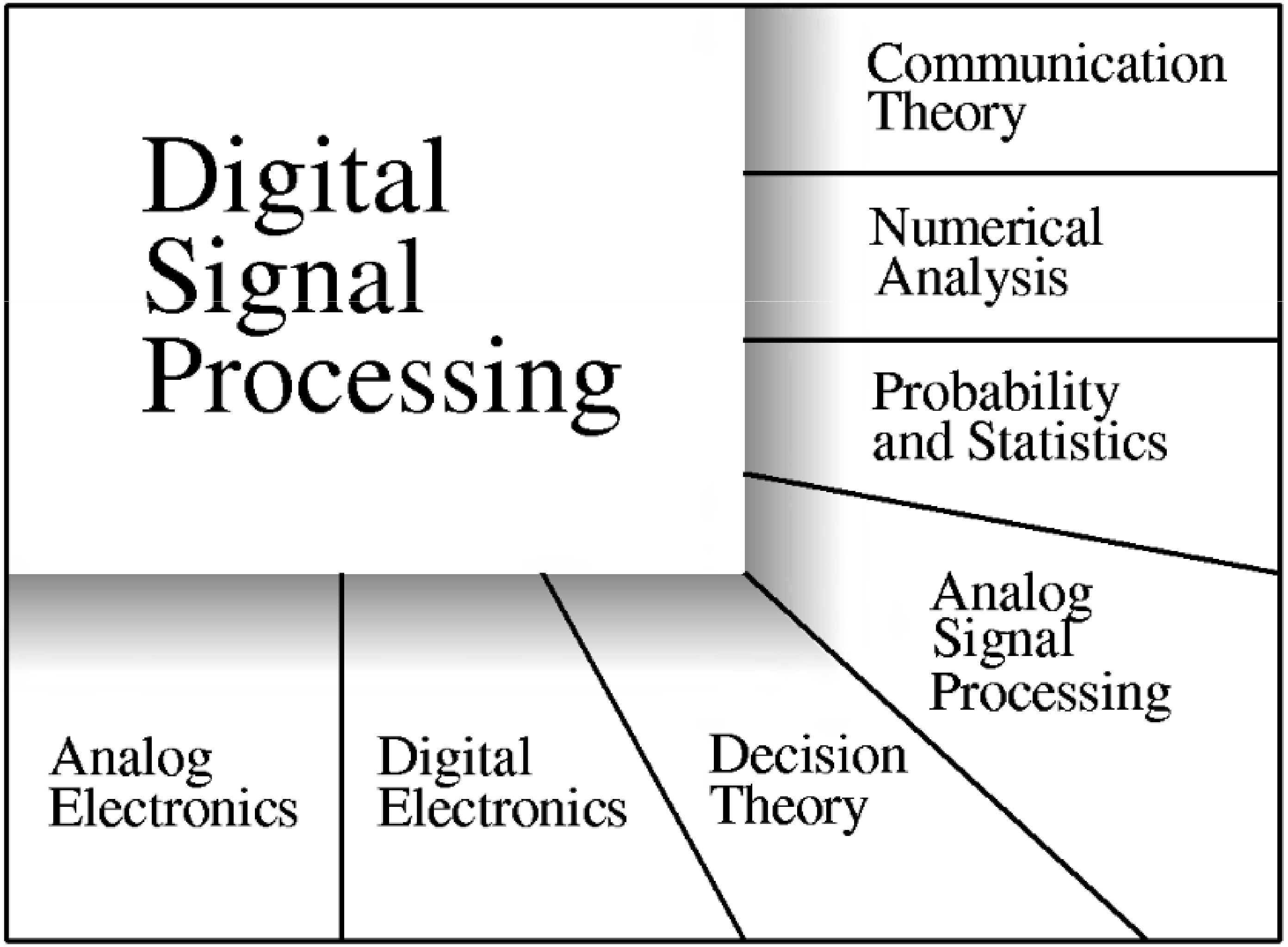
What is DSP

- The mathematics, algorithms, and techniques used to manipulate signals after converting them to digital form (Smith)
- Anything that can be done to a signal using code on a computer or DSP chip (White)

Examples of DSP



Relation to Other Fields



DSP Magic

- Telecommunication
  - Multiplexing

    T-carrier systems transmit 24 voice signals (1.544Mbit/sec)
  - Compression

    64Kbit/sec sound (8000*8) → 2Kbit/sec
  - Echo control

    Antisignal generation

DSP Magic cont.

- Audio Processing
  - Music
    artificial reverberation
  - Speech Generation
    Digital Recording
    Vocal tract simulation
  - Speech Recognition
    Still far far away
  - Speaker Localization
    Microphone array

DSP Magic cont.

- Echo Location
  - Radar (Radio Detection And Ranging)
    energy-noise dependence, energy-length tradeoff
    > 100 MHz real-time speed
  - Sonar (SOund Navigation And Ranging)
    active (2K-40K)
    passive(military)
    pulse generation, compression and noise filtering
  - Reflection Seismology
    echo of echo problem

DSP Magic cont.

- Image Processing
  - Medical

    from X ray to Computed Tomography (Nobel Prize 79)
  - Space

    Image enhancement
  - Commercial Products

    image compression

    video codecs

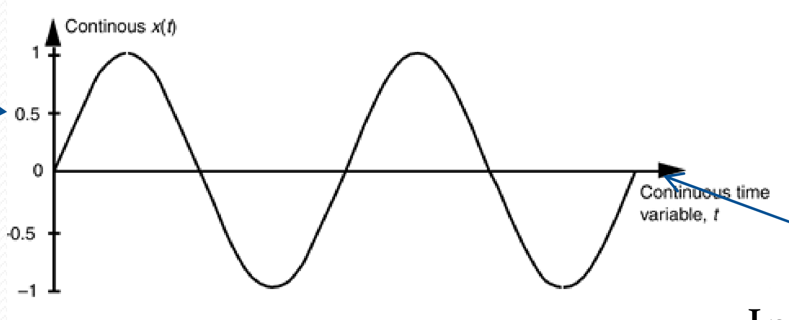
    face detection

    face recognition

What is a signal

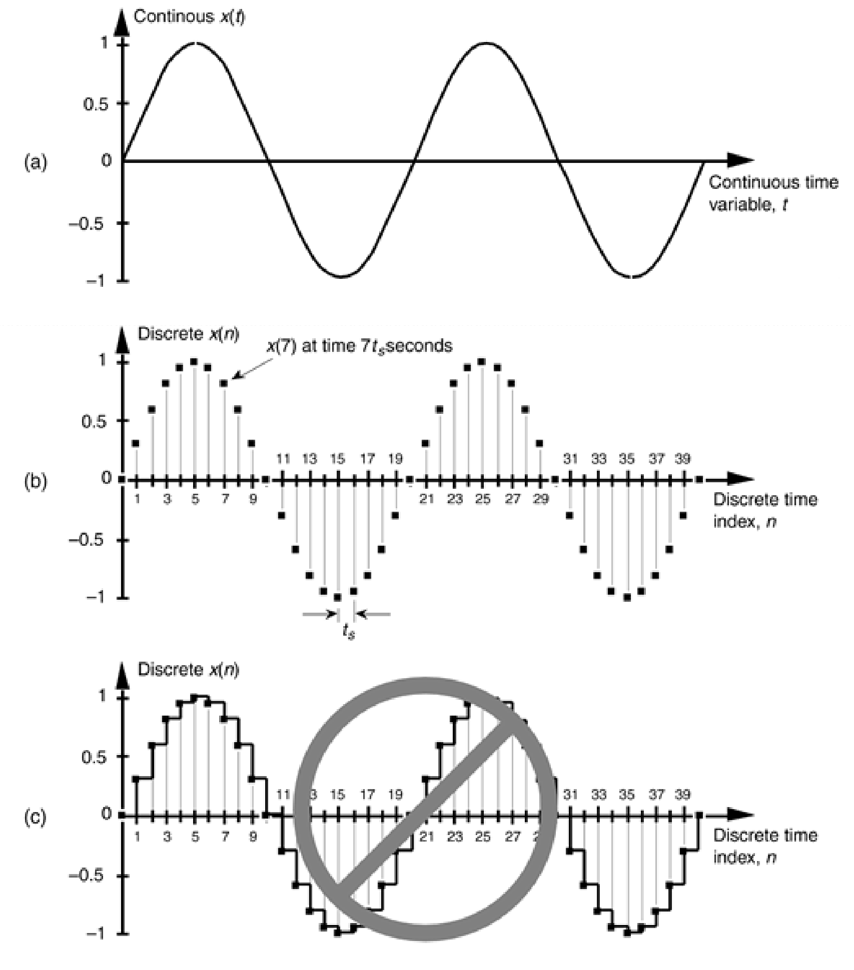
- A description of how one (or more) value is changing depending on another (one or more) parameters
- Examples
  - sin(t)
  - 1,0.343,45,4343,49.032,-343.0, ...

Dependent variable



Independent variable

What is a discrete signal



Continuous Signal $x(t) = \sin(2\pi f_o t)$.

Discrete Signal $x(n) = \sin(2\pi f_o n t_s)$

????? Signal

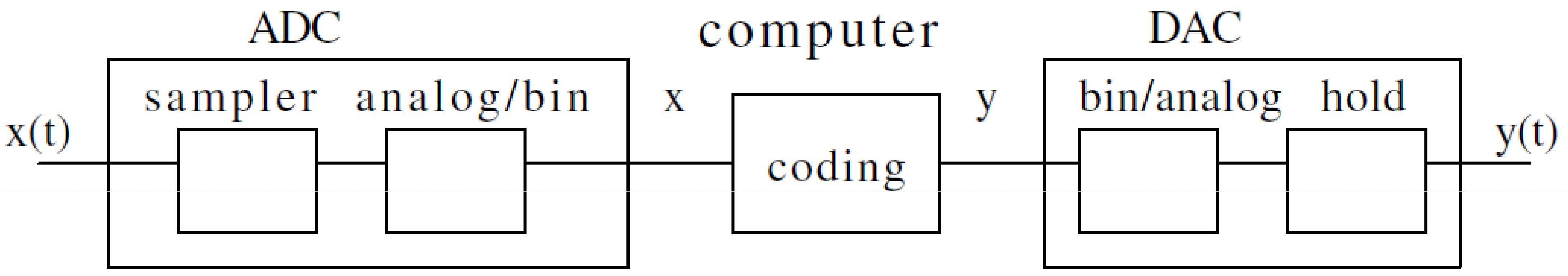
Types of Signals

		Independent	
Dependent		Discrete	Continuous
	Discrete	Quantized Discrete Signal (Digital Signal) $x[n]$	Quantized Signal $\bar{x}(t)$
	Continuous	Discrete or Digitized Signal $\tilde{x}[n]$	Continuous or Analog Signal $x(t)$

Note on notation

- x[n] sample number *n* of a discrete signal (may be quantized or not)
- x_k x[n+k] step number k from current step.

General DSP System



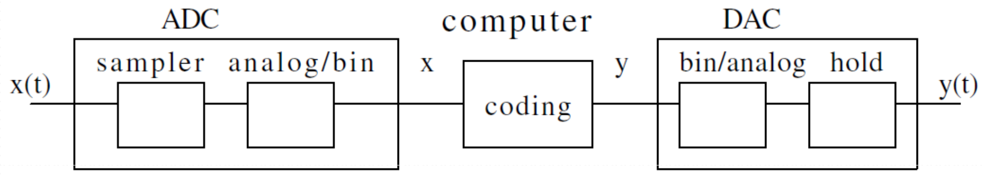
$$y = (b_{-1}y_{-1} + \cdots + b_{-m}y_{-m}) + (ax + a_{-1}x_{-1} + \cdots + a_{-n}x_{-n})$$

$$y = \sum_{i=1}^{m} b_{-i}y_{-1} + \sum_{j=0}^{n} a_{-j}x_{-1}$$

Nearly ALL DSP that you need is contained in this equation

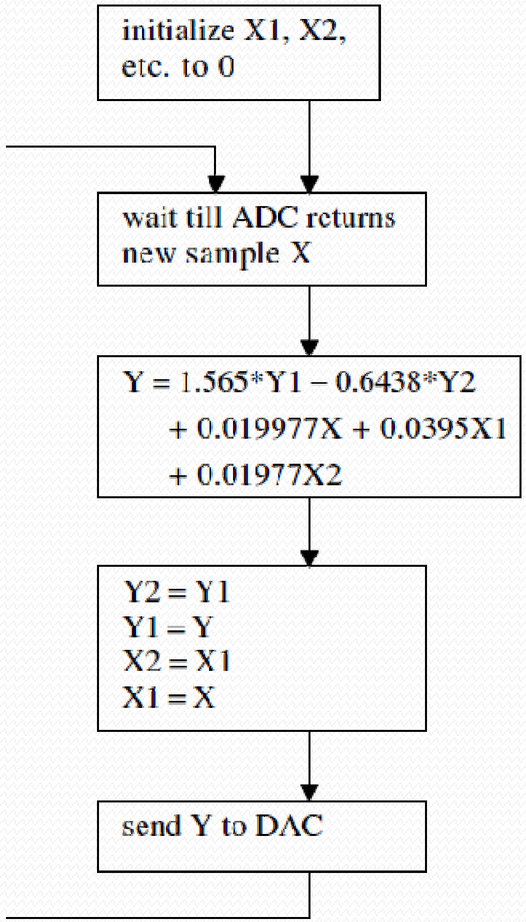
Simplest DSP systems

- $y[n]=0$ (Broken DSP system!!)
- $y[n]=x[n]$ (Buffer)



- $y[n]=x[n-k]$ (Delay k steps)
- $y[n]=a.x[n]$ (Amplifier [$a>1$] /Attenuator [$a<1$])
- $y[n]=x[n]+y[n-1]$

Simple Example



X1, X2, ... are names for x_{-1}, x_{-2}, etc.

X is name for x

Y is name for y, Y1 is name for y_{-1}, etc.

This is the coded DSP equation

Saving previous values, only 2 shown here

First Assignment (In lecture)

1. Change the equations for the following signals to describe the signals after they go through an ADC with a sample period of T seconds.

   (a) $x(t) = e^{-3t}$

   (b) $x(t) = 5t^2$

1a. $x(n) = e^{-3nT}$

1b. $x(n) = 5(nT)^2$

2. Compute the value of the sample for $n = 10$ for the following signals after they have gone through an ADC with the sample time $T = 0.05$ seconds.

   (a) $x(t) = 7\sin(25t)$

   (b) $x(t) = 2\cos(50t) - 4\cos(100t)$

2a. $x(10) = -0.464$

2b. $x(10) = -1.877$

3. Compute the values of the following signals after going through an ADC with $T = 0.1$ s for the values of n from 0 to 10.

   (a) $x(t) = 2\cos(10t)$

$x(0) = 2.0,\ x(1) = 1.08$

Domains

- Domain: Type of the independent parameter (ind)
- Most Useful domains:
  - Time Domain (ind = time, sample number, space)
  - Frequency Domain (ind = frequency)
    - Fourier (series, transform), DFT
    - Laplace transform
    - Z- transform
  - Time-Frequency Domain (ind=frequency+time)
    - Wavelet transform
    - Short-time Fourier transform
- In most of this course we will iterate between time and frequency domains.
- Time Frequency representation (TFR) is still an advanced topic in DSP