



## BLM3620 Digital Signal Processing

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## Lecture #1 – Introduction to DSP

- General Information about the Course
- Recommended Documents
- Introduction
- DSP Applications
- Basic Signal Operations

## Course Materials

### Important Materials:

- James H. McClellan, R. W. Schafer, M. A. Yoder, *DSP First Second Edition*, Pearson, 2015.
- Lizhe Tan, Jean Jiang, *Digital Signal Processing: Fundamentals and Applications*, Third Edition, Academic Press, 2019.

### Auxiliary Materials:

- Prof. Sarp Ertürk, *Sayısal İşaret İşleme*, Birsen Yayınevi.
- Prof. Nizamettin Aydın, DSP Lecture Notes.
- J. G. Proakis, D. K. Manolakis, *Digital Signal Processing Fourth Edition*, Pearson, 2014.
- J. K. Perin, *Digital Signal Processing, Lecture Notes*, Stanford University, 2018.

## General Information

Ders Adı	Kodu	Yerel Kodu	AKTS	Ders Saati (hafta)	Uygulama Saati (hafta)	Laboratuvar Saati (hafta)
Sayısal İşaret İşleme	BLM3620	3	5	3	0	0

Değerlendirme Sistemi		
Etkinlikler	Sayı	Katkı Payı
Devam/Katılım		
Laboratuvar		
Uygulama		
Arazi Çalışması		
Derse Özgü Staj		
Küçük Sınavlar/Stüdyo Kitabı		
Ödev	4	30
Sorumlu		
Projeler		
Seminer/Workshop		
Ara Sınavlar	1	30
Final	1	40
Dönem İçi Çalışmaların Başarı Notuna Katkısı		60
Final Sınavının Başarı Notuna Katkısı		40

## Course Materials

James H. McClellan, R. W. Schafer, M. A. Yoder, *DSP First Second Edition*, Pearson, 2015.

<https://dspfirst.gatech.edu/>

- Demos,
- Examples,
- Exercises,
- Lectures,
- Labs...



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## Syllabus

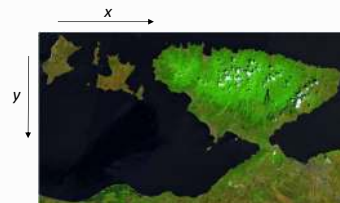
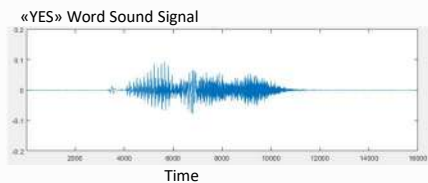
Week	Lectures
1	Introduction to DSP and MATLAB
2	Sinusoids and Complex Exponentials
3	Spectrum Representation
4	Sampling and Aliasing
5	Discrete Time Signal Properties and Convolution
6	Convolution and FIR Filters
7	Frequency Response of FIR Filters
8	Midterm Exam
9	Discrete Time Fourier Transform and Properties
10	Discrete Fourier Transform and Properties
11	Fast Fourier Transform and Windowing
12	z-Transforms
13	FIR Filter Design and Applications
14	IIR Filter Design and Applications
15	Final Exam

For more details -> Bologna page: <http://www.bologna.yildiz.edu.tr/index.php?r=course/view&id=5730&aid=3>

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## COURSE OBJECTIVE

- Students will be able to:
- Understand **mathematical** descriptions of signal processing **algorithms** and express those algorithms as computer **implementations** (MATLAB)
- What are your objectives?



Mathematical Function:  $I(x, y)$

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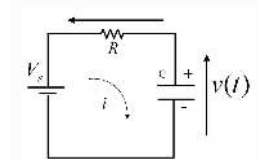
## What are Signals and Systems?

### ► Signal:

- any physical quantity that varies with time, space, or any other independent variable or variables
- Examples: pressure as a function of altitude, sound as a function of time, color as a function of space, ...
- $x(t) = \cos(2\pi t)$ ,  $x(t) = 4\sqrt{t} + t^3$ ,  $x(m, n) = (m + n)^2$

### ► System:

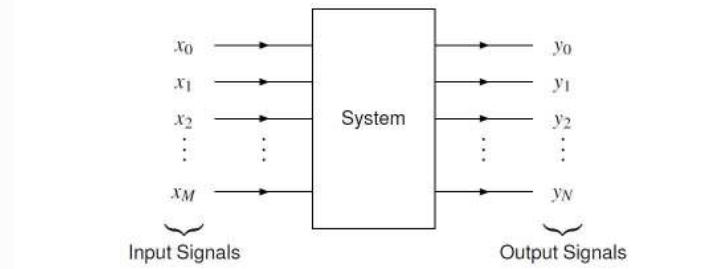
- a physical device that performs an operation on a signal
- Examples: analog amplifier, noise canceler, communication channel, transistor, ...
- $y(t) = -4x(t)$ ,  $\frac{dy(t)}{dt} + 3y(t) = -\frac{dx(t)}{dt} + 6x(t)$ ,  
 $y(n) - \frac{1}{2}y(n-2) = 3x(n) + x(n-2)$



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## What is a system?

A **system** is an entity that processes one or more input signals in order to produce one or more output signals.

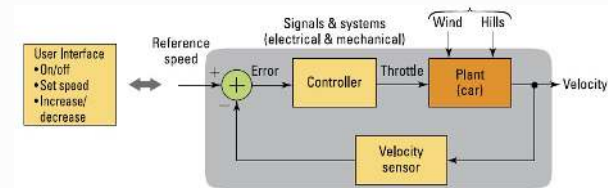


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## Example: Cruise Control System

Input Signals: ?

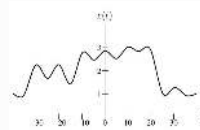
Output Signals: Speed of car



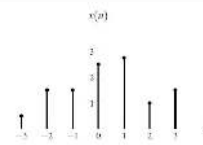
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## Classification of Signals

- Number of independent variables (i.e., dimensionality):
  - A signal with **one** independent variable is said to be **one dimensional** (e.g., audio).
  - A signal with **more than one** independent variable is said to be **multi-dimensional** (e.g., image).
- Continuous or discrete independent variables:
  - A signal with **continuous** independent variables is said to be **continuous time (CT)** (e.g., voltage waveform).
  - A signal with **discrete** independent variables is said to be **discrete time (DT)** (e.g., stock market index).



Continuous-Time (CT) Signal



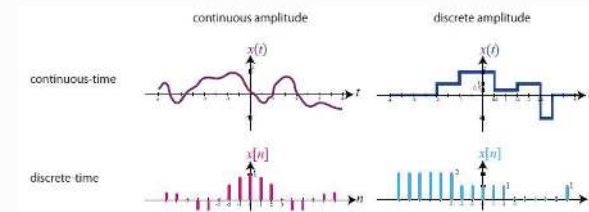
Discrete-Time (DT) Signal

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## Understanding Analog and Digital Signals

**Analog signal** -> continuous both in time and amplitude

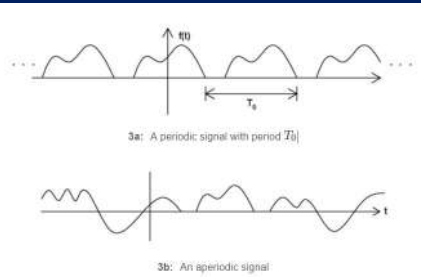
**Digital signal** -> discrete both in time and amplitude



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## Classifications of Signals

### Periodic vs. Aperiodic



Periodic signals repeat with some **period**  $T$ , while aperiodic, or nonperiodic, signals do not. We can define a periodic function through the following mathematical expression, where  $t$  can be any number and  $T$  is a positive constant

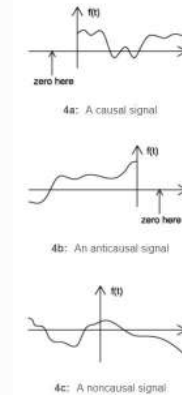
$$f(t) = f(t + T)$$

**fundamental period** of our function,  $f(t)$ , is the smallest value of  $T$  that the still allows Equation to be true

## Classifications of Signals

### Causal vs. Anticausal vs. Noncausal

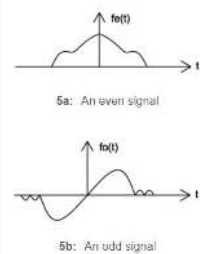
- **Causal** signals are signals that are zero for all negative time, while **anticausal** are signals that are zero for all positive time.
- **Noncausal** signals are signals that have nonzero values in both positive and negative time



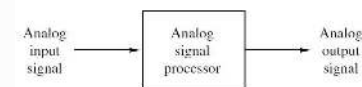
## Classifications of Signals

- An **even signal** is any signal  $f$  such that  $f(t)=f(-t)$ . Even signals can be easily spotted as they are **symmetric** around the vertical axis.
- An **odd signal**, on the other hand, is a signal  $f$  such that  $f(t)=-f(-t)$

### Even vs. Odd



## Analog Systems vs. Digital Systems



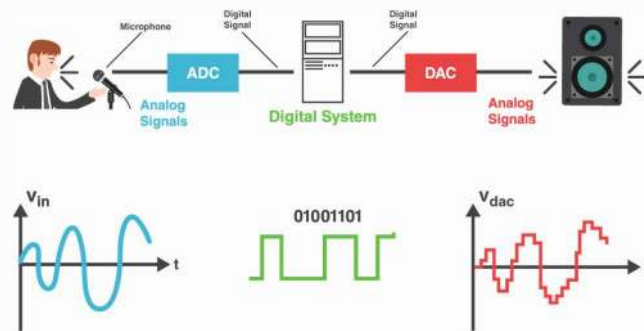
### Analog Systems:

- Directly use real-world signals.
- Do not need to an ADC or a DAC.
- Give the fastest application results.

### Digital Systems:

- have lower distortions thanks to error correction.
- Digital signals can easily be compressed and saved.
- Do not include any R, L, C elements. (programmed on software)
- Are more stable and robust to the environmental conditions.
- Can be ported different hardware

## General Concept in DSP Applications

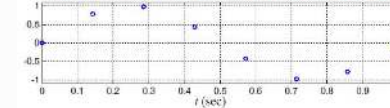


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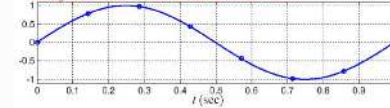
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## Sampling is very important !

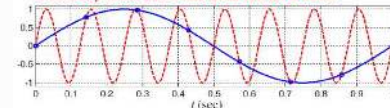
DT Samples.... What CT Sinusoid did they come from????



They could have come from this blue one...



But... They could have come from this RED one!!!



Thus... if we want to be able to tell these two apart we need to sample faster!!

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## TUNING FORK EXAMPLE

- CD-ROM demo
- "A" is at 440 Hertz (Hz)
- Waveform is a SINUSOIDAL SIGNAL
- Computer plot looks like a sine wave
- This should be the mathematical formula:



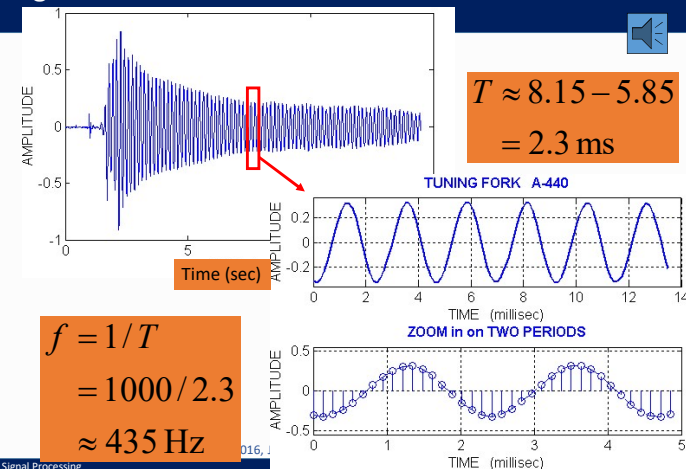
$$A \cos(2\pi(440)t + \varphi)$$

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## Tuning Fork: A-440 Waveform



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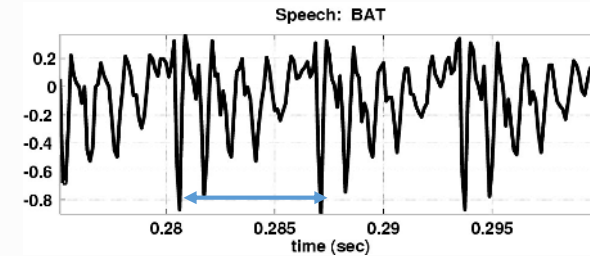
## What about speech?

- More complicated signal (BAT.WAV)
- Waveform  $x(t)$  is NOT a Sinusoid
- Theory will tell us
  - $x(t)$  is approximately a sum of sinusoids
  - FOURIER ANALYSIS
    - Break  $x(t)$  into its sinusoidal components
  - Called the FREQUENCY SPECTRUM



## What about speech?

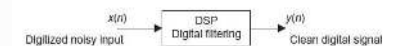
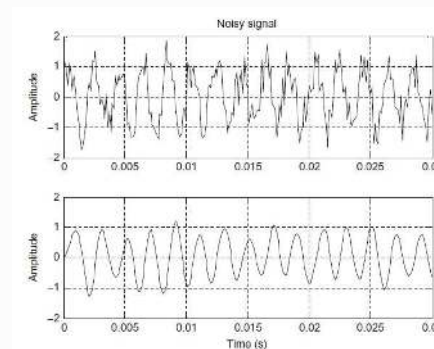
- Nearly **Periodic** in Vowel Region
  - Period is (Approximately)  $T = 0.0065$  sec



## STORING DIGITAL SOUND

- $x[n]$  is a SAMPLED SINUSOID
  - A list of numbers stored in memory
- CD rate is 44,100 samples per second
- 16-bit samples
- Stereo uses 2 channels
- Number of bytes for 1 minute is
  - $2 \times (16/8) \times 60 \times 44100 = 10.584$  Mbytes

## DSP Applications: Digital Filtering



Which filter type should we use?

## DSP Applications: Digital Crossover Audio System

Analog input → Analog filter → ADC →  $x(n)$  → DSP Algorithms → Time domain display / Frequency content display

We should go to frequency domain...

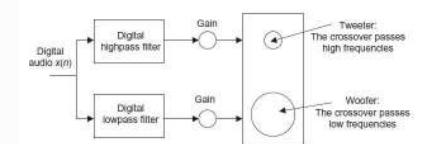
(A) Signal amplitude vs. Time (s). (B) Signal spectrum vs. Frequency (Hz). (C) Signal amplitude vs. Time (s). (D) Signal spectrum vs. Frequency (Hz).

Ref. J. Jiang and L. Tan

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## Some Real-World Apps: Digital Crossover Audio System

The incoming digital audio signal is split into two bands using a digital lowpass filter and a digital highpass filter in parallel. Then the separated audio signals are amplified. Finally, they are sent to their corresponding speaker drivers.



Although the traditional crossover systems are designed using the analog circuits, the digital crossover system offers a cost-effective solution with programmable ability, flexibility, and high quality.

Ref. J. Jiang and L. Tan, DSP: Fundamentals and Apps

## Some Real-World Apps: Filtering ECG Signal

Design a notch filter and apply it to digital signal

ECG signal with 60-Hz interference → ECG preamplifier → Digital notch filter for eliminating 60-Hz interference → ECG recorder with the removed 60-Hz interference

Ref. J. Jiang and L. Tan, DSP: Fundamentals and Apps

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## Some Real-World Apps: Vibration Signature for Defect Det.

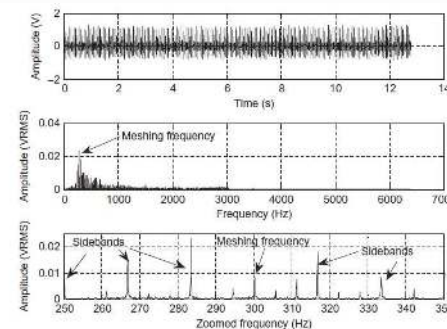
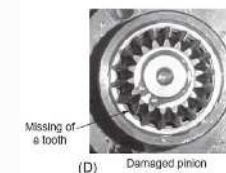


FIG. 1.13  
Vibration signal and spectrum from the damaged gearbox.



Ref. J. Jiang and L. Tan, DSP: Fundamentals and Apps



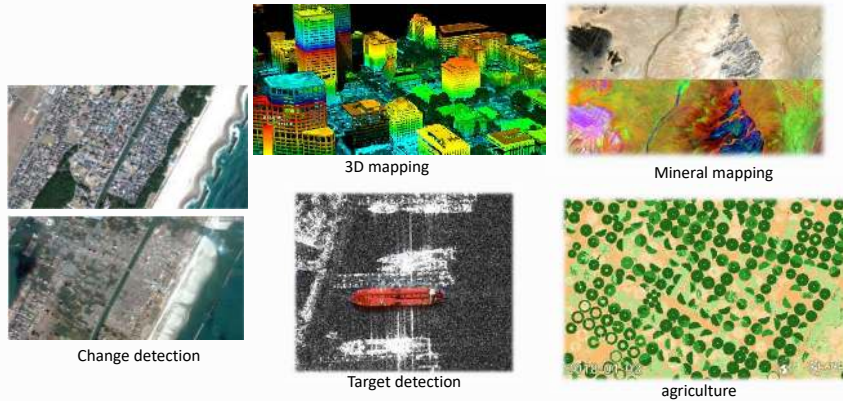
## Other Applications

DSP = Swiss-Army-Knife of modern CEng

- Communications (wireless, internet, GPS),
- Control and monitoring (cars, machines...),
- Multimedia (videos, cameras, ...),
- Healthcare (medical devices),
- More...



## Image Processing – Remote Sensing



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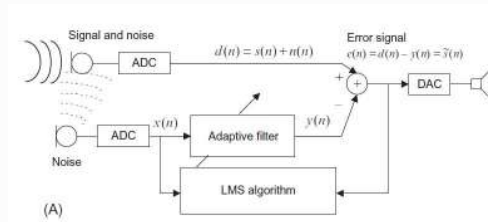
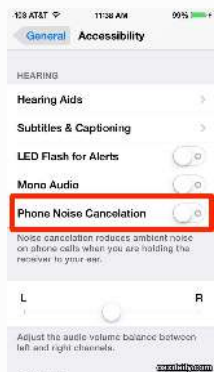
## Some Real-World Apps: Autotune



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## Some Real-World Apps: Noise Cancellation



Ref. J. Jiang and L. Tan, DSP: Fundamentals and Apps

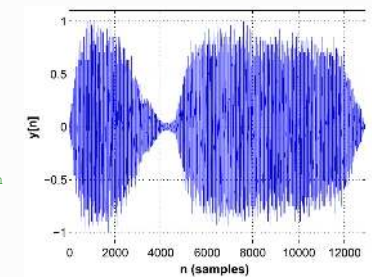
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## Let's take a look at train signal...

```
clear all;
load train;

%% Example---Listening to/plotting train signal
sound(y,Fs)
t=0:1/Fs:(length(y)-1)/Fs;
figure(2); plot(t,y'); grid
ylabel('y[n]'); xlabel('n')

%% Example---Using stem to plot 200 samples of train
figure(3)
n=100:299;
stem(n,y(100:299)); ylabel('y[n]'); xlabel('n')
title('Segment of train signal')
axis([100 299 -0.5 0.5])
```



How can we decrease the amplitude of the sound??

What about noisy signals??

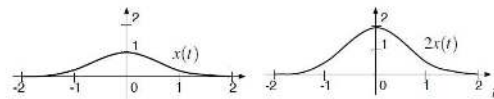
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## Operations on Signals

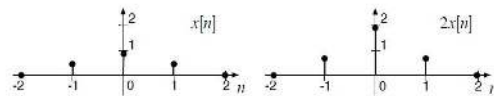


### Amplitude Scaling

- The scaled signal  $ax(t)$  is  $x(t)$  multiplied by the constant  $a$



- The scaled signal  $ax[n]$  is  $x[n]$  multiplied by the constant  $a$



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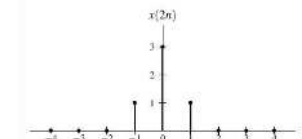
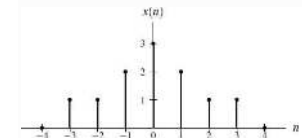
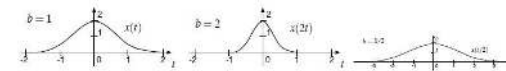
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## Operations on Signals



### Time Scaling, Continuous Time

A signal  $x(t)$  is scaled in time by multiplying the time variable by a positive constant  $b$ , to produce  $x(bt)$ . A positive factor of  $b$  either expands ( $0 < b < 1$ ) or compresses ( $b > 1$ ) the signal in time.



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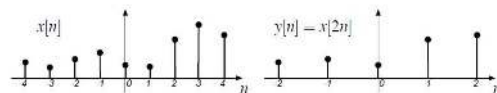
## Operations on Signals



### Time Scaling, Discrete Time

The discrete-time sequence  $x[n]$  is *compressed* in time by multiplying the index  $n$  by an integer  $k$ , to produce the time-scaled sequence  $x[nk]$ .

- This extracts every  $k^{\text{th}}$  sample of  $x[n]$ .
- Intermediate samples are lost.
- The sequence is shorter.



Called *downsampling*, or *decimation*.

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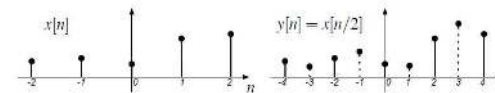
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## Operations on Signals



The discrete-time sequence  $x[n]$  is *expanded* in time by dividing the index  $n$  by an integer  $m$ , to produce the time-scaled sequence  $x[n/m]$ .

- This specifies every  $m^{\text{th}}$  sample.
- The intermediate samples must be synthesized (set to zero, or interpolated).
- The sequence is longer.



Called *upsampling*, or *interpolation*.

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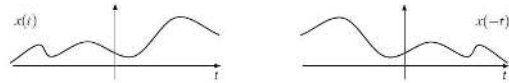
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## Operations on Signals



### Time Reversal

- Continuous time: replace  $t$  with  $-t$ , time reversed signal is  $x(-t)$



- Discrete time: replace  $n$  with  $-n$ , time reversed signal is  $x[-n]$



- Same as time scaling, but with  $b = -1$ .

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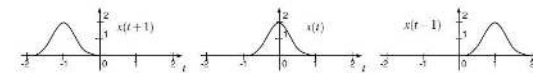
## Operations on Signals



### Time Shift

For a continuous-time signal  $x(t)$ , and a time  $t_1 > 0$ ,

- Replacing  $t$  with  $t - t_1$  gives a *delayed* signal  $x(t - t_1)$
- Replacing  $t$  with  $t + t_1$  gives an *advanced* signal  $x(t + t_1)$



- May seem counterintuitive. Think about where  $t - t_1$  is zero.

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## Operations on Signals

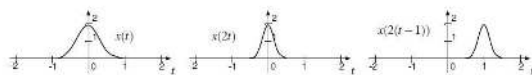


### Combinations of Operations

- Time scaling, shifting, and reversal can all be combined.
- Operation can be performed in any order, but care is required.
- This *will* cause confusion.
- Example:  $x(2(t-1))$

Scale first, then shift  
Compress by 2, shift by 1

2 \* 2



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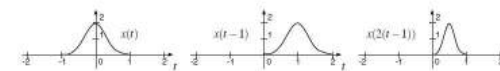
## Example-1



Example  $x(2(t-1))$ , continued

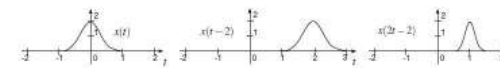
Shift first, then scale  
Shift by 1, compress by 2

*Incorrect*



Shift first, then scale  
Rewrite  $x(2(t-1)) = x(2t-2)$   
Shift by 2, scale by 2

*Correct*

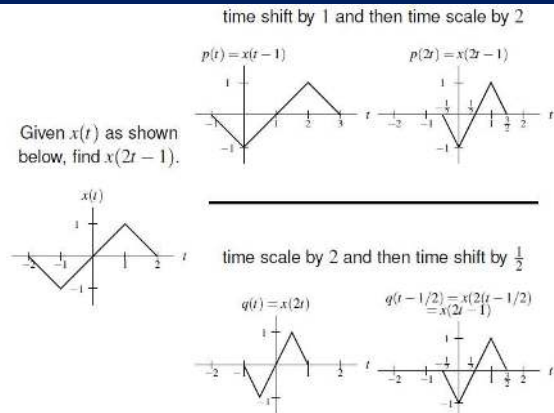


Where is  $2(t-1)$  equal to zero?

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## Example-2

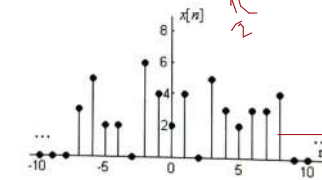


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## Homework (Just for you, do not e-mail)

## Problemler:

2.1.  $x[n]$  dizisi Şekil 2.23'de gösterilmektedir. Aşağıdaki dizileri çiziniz.(a)  $y[n] = x[n-3]$  (b)  $y[n] = x[-n] - x[n+2]$  (c)  $y[n] = x[-n+3]$  (d)  $y[n] = x[6-3n]$ (e)  $y[n] = x[n/2-4]$ Şekil 2.23. Problem 2.1. için  $x[n]$  işareti

@Sayısal İşaret İşleme, Sarp ERTÜRK

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## Delta Function and Unit Step Function (Please Do Not Forget it!)

Unit Step Function

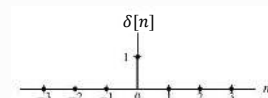
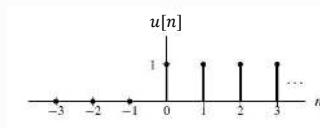
$$u(n) = \begin{cases} 1 & \text{if } n \geq 0 \\ 0 & \text{otherwise} \end{cases}$$

Unit Impulse Function

$$\delta(n) = \begin{cases} 1 & \text{if } n = 0 \\ 0 & \text{otherwise} \end{cases}$$

Relations

$$\delta[n] = u[n] - u[n-1]$$



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## Plot the signals given below:

1)  $x[n] = 2\delta[n-1] + 3\delta[n-2] + 4\delta[n-3]$  işaretini çiziniz.2)  $y[n] = 3\delta[n-2] + u[n-5] - u[n-7]$  işaretini çiziniz.3)  $g[n] = y[n]x[n]$  ifadesini bulunuz.4)  $h[n] = y[n] + x[n]$  ifadesini bulunuz.

MATLAB kodu:

```
clc; clear all;
%%
n = [0 1 2 3 4 5 6 7];
x = [0 2 3 4 0 0 0 0];
figure(1), stem(n,x,'filled');
```

```
%%
n = [0 1 2 3 4 5 6 7];
y = [0 0 3 0 0 1 1 0];
figure(2), stem(n,y,'filled');
```

```
%%
g=y.*x;
figure(3), stem(n,g,'filled');
```

```
%%
h=y+x;
figure(4), stem(n,h,'filled');
```

BLM3620 Digital Signal Processing

