Al-Farahidi University

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الكلية التقنية الهندسية
هندسة تقنيات الاتصالات

المرحلة الثالثة
معالج اشارة رقمية

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Third year
2023-2024

DIGITAL SIGNAL PROCESSING

Reference

1. Digital Signal Processing Principles, Algorithms, and Applications fourth Edition

John G. Proakis Northeastern University Dimitris

G. Manolakis Boston College

2. Schaum's Outline of Theory and Problems of Digital Signal Processing

Monson H. Hayes Professor of Electrical and Computer Engineering Georgia Institute of Technology

SCHAUM'S OUTLINE SERIES

3. Digital signal processing
V.K. KHANNA

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Third year
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Lecture One

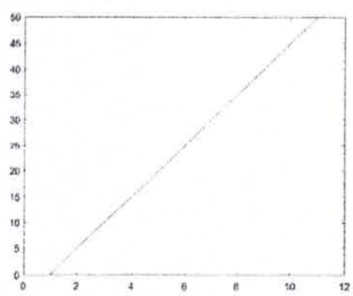
INTRODUCTION TO DIGITAL SIGNAL PROCESSING

What is Signal?

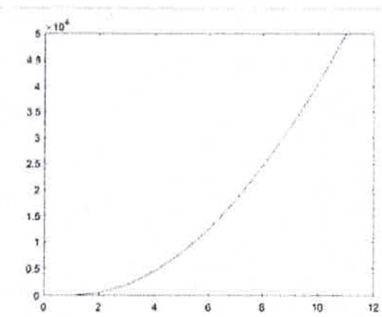
- *A **signal** is defined as any physical quantity that varies with*
  - *time,*
  - *space, or*
  - *any* other independent variable or variables.
- Mathematically, a signal is a function of one or more independent variables.

Examples of signals

- **e.g. of signals**
  - $s_1(t) = 5t$, *(linearly)*
  - $s_2(t) = 20t^2$ *(quadratic)*
  - $s(x, y) = 3x + 2xy + 10y^2$
  - $\sum_{i=1}^{N} A_i(t) \sin[2\pi F_i(t)t + \theta_i(t)]$ (speech signal)



s1



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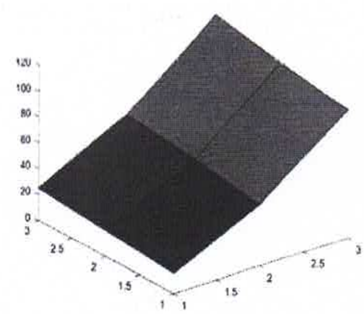
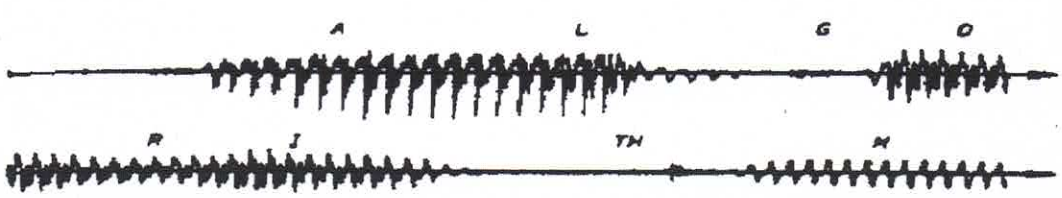


s(x,y)



What is a System?

- *A system may also be defined as a physical device that performs an operation on a signal.*
  - For example, a filter used to reduce the noise and interference corrupting a desired information-bearing signal is called a system.

What is signal processing?

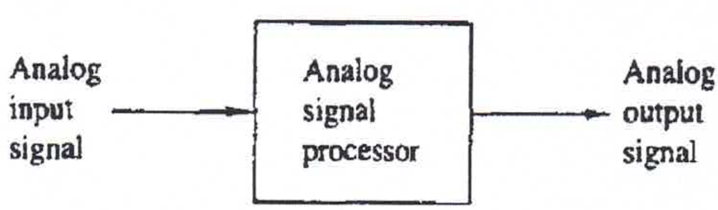
- When a signal is passed through a system, as in filtering, this means that the signal was processed.
- Such operations are usually referred to as *signal processing.*
- In general, the system is characterized by the type of operation that it performs on the signal.
  - For example, if the operation is linear, the system is called linear.

Why signals should be processed?

- Signals are carriers of information
  - Useful and unwanted
  - Extracting, enhancing, storing and transmitting the useful information
- How signals are being processed? ---
  - Analog Signal Processing vs.
  - Digital Signal Processing

Analog Signal Processing

- Signals may be processed directly by appropriate analog systems (such as filters or frequency analyzers) or frequency multipliers for the purpose of changing their characteristics or extracting some desired information.

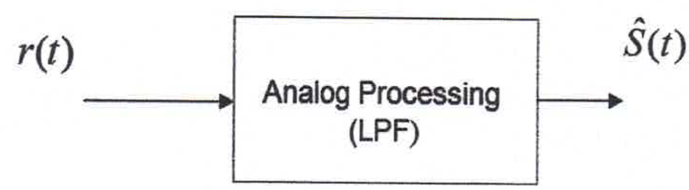


Example of ASP

- Let s(t) be the transmitted signal and r(t) be the received signal.
- The received signal will contain both the useful signal s(t) and the unwanted signal in the form of noise n(t). Thus;

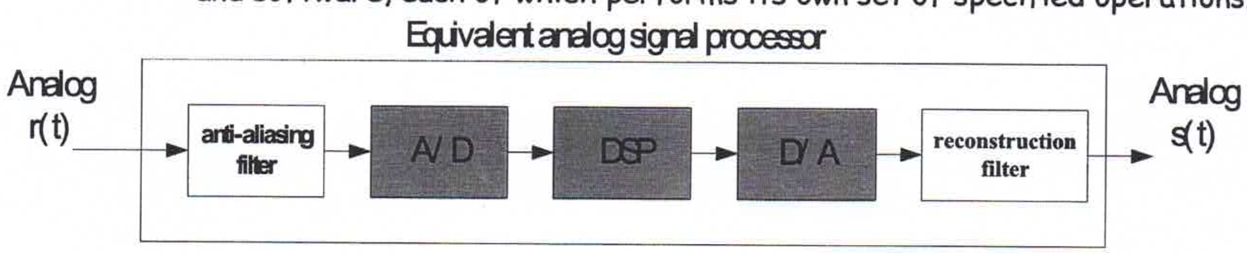
$$r(t) = s(t) + n(t)$$

- The main function at the receiver would be to extract the useful signal s(t), or obtain an estimate of the useful signal from the received signal r(t).
- Usually the noise is of high frequency and so the analog processor could be an analog low pass filter (LPF) which will reduce the effect of high frequency noise and extract the useful signal component.



Digital Signal Processing

- A digital system can be implemented as a combination of digital hardware and software, each of which performs its own set of specified operations.



- The first stage of DSP system is usually a **pre-filter (anti-aliasing filter)**
  - The filter limits the frequency band of the input signal and hence reduces the possible interference.
- This is followed by an **analog-to-digital converter (A/D or ADC)** which converts the analog signal to a digital signal.
  - This actually consists of sampling operation followed by amplitude quantization.
- The digital output of A/D converter is then processed by a **Digital Processor** which could be a digital filter or any other digital operation.

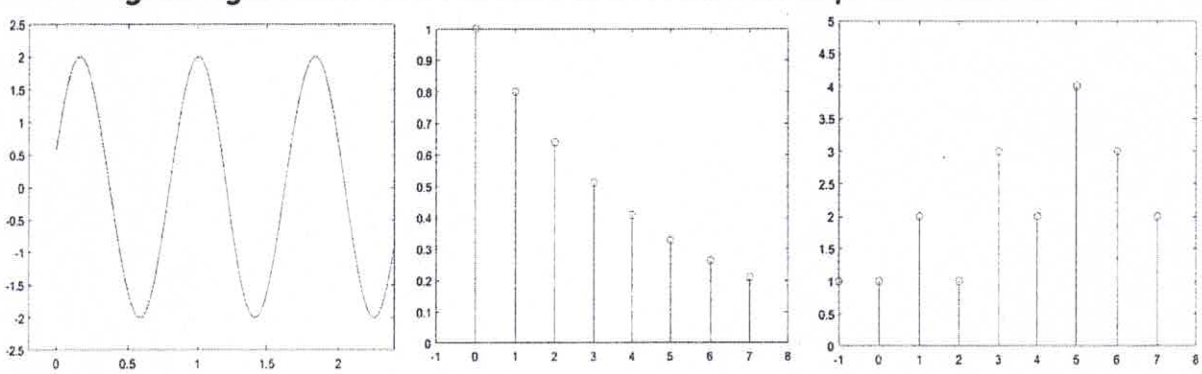
- The processed digital signal is then converted back to analog form by a **Digital-to-Analog converter (D/A or DAC).**
- The output of the DAC has a staircase wave form. This is then smoothened by a **reconstruction filter** to get an estimate of the wanted signal s(t).

Here we have replaced the simple analog filter with a pre-filter, ADC, DAC, digital processor, and a reconstruction filter. So, what is the advantage that we have gained?

- **We have gained flexibility** at the expense of complexity.
- Suppose we want to change some parameter of the system, for example the cutoff frequency of the filter.
    - In this case the change is made by reprogramming of the digital processor (change of the coefficients of the digital filter) only, without changing the actual structure. So, there is a great deal of flexibility with DSP.

Continuous, Discrete & Digital Signals

- **Continuous time (CT) or analog signal** signals are both continuous in the time and amplitude.
    - **Analog sinusoidal signal.**
- **Discrete time (DT)** signals are discrete in the time domain but continuous in the amplitude.
    - **Discrete time signal** *$x[n]=0.8^n$ for $n > 0$ and $x(n) = 0$ for $n < 0$.*
- **Digital signals** are both discrete in the time and amplitude domains.



CT DT DS

The Sampling Process

- During the process of sampling, the CT signal is sampled at certain time instants.
- The duration between any successive instants is called the sampling interval **t_s**

- The inverse of the sampling interval t_s (in seconds) is called the sampling frequency f_s (in hz or samples/s).
- $\frac{1}{t_s} = f_s$

The Sampling Theorem

The sampling theorem states that;

- The sampling frequency f_s should be greater than or equal to twice the maximum frequency of the CT signal f_{max} (i.e. $f_s \geq 2f_{max}$) in order to preserve the information of the CT signal.

Nyquist's Sampling Rate

- The minimum value of the sampling frequency according to the sampling theorem is called the *Nyquist's Sampling Rate*. **i.e.** $f_{Nyq} = 2f_{max}$

Example 1.4.3

Consider the analog signal

$$x_a = 3cos50\pi t + 10sin300\pi t - \cos 100\pi t$$

what is the Nyquist rate for the signal?
Solution:

$$2\pi F = 50\pi, 2\pi F = 300\pi, 2\pi F = 100\pi$$

$$F_1 = 25 \text{ Hz}, \quad F_2 = 150 \text{ Hz}, \quad F_3 = 50 \text{ Hz}$$

$F_{max}=150$, since $F_s \geq 2F_{max} = 300\ Hz = F_N$ the Nyquist rate

See Example 1.4.4 page 31, reference 1

The Quantization process

- A signal in the time domain is made discrete in the amplitude domain by the Quantization process.
- In quantization, the amplitude of the signal is approximated to standard M quantization levels.
- These standard levels are encoded to produce digital signal. Usually $M=2^n$, where n and M are integers & n is the number of bits per sample.
- The value of n is called the signal resolution in most applications of DSP.
  - To process analog signals by digital means, it is first necessary to convert them into digital form, that is, to convert them to a sequence of numbers

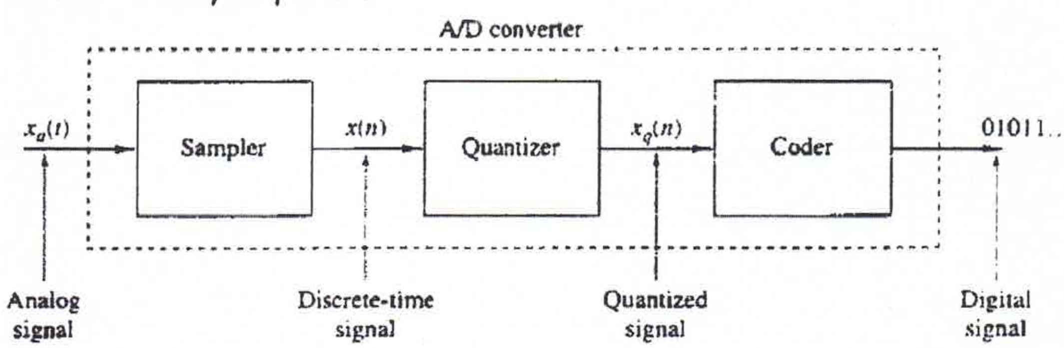
having finite precision. This procedure is called *analog-to-digital (A/D) conversion*,

- A /D conversion has a three-step process:

1. *Sampling*. This is the conversion of a continuous-time signal in to a discrete time signal obtained by taking samples of the continuous-time signal at discrete-time instants. Thus, if $x_a(t)$ is the input to the sampler, the output is $x_a(nT) = x(n)$, where T is called the sampling interval.

- The time interval T between successive samples is called the sampling period or sample interval and its reciprocal $1/T = F_s$ is called the sampling rate (samples per second) or the sampling frequency (hertz).

2. *Quantization*. This is the conversion of a discrete-time continuous-valued signal in to a discrete-time, discrete-valued (digital) signal. The value of each signal sample is represented by a value selected from a finite set of possible values. The difference between the un-quantized sample x(n) and the quantized output $x_q(n)$ is called the quantization error.
3. *Coding*. In the coding process, each discrete value $x_q(n)$ is represented by a b-bit binary sequence.



Comparison of DSP over ASP

-Advantages

- Developed Using Software on Computer;
- Working Extremely Stable;
- Easily Modified in Real Time;
- Low Cost and Portable;

-Disadvantages

- Lower Speed and Lower Frequency

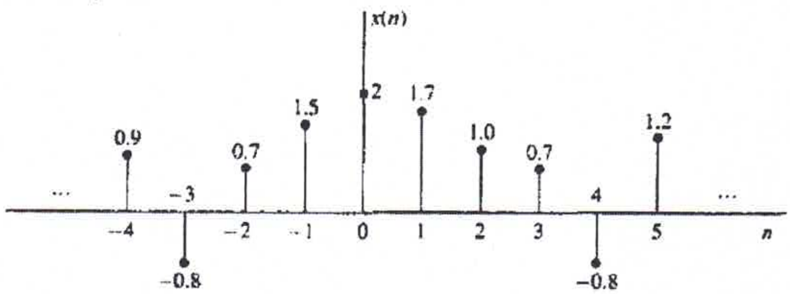
The two categories of DSP Tasks

- Signal Analysis:
  - Measurement of signal properties
  - Spectrum(frequency/phase) analysis
  - Target detection, verification, recognition
- Signal Filtering
  - Signal-in-signal-out, filter
  - Removal of noise/interference
  - Separation of frequency bands

Signal Representation

- A Sampled or discrete times signal x(n) is just an ordered sequence of values corresponding to the index n that indicates the time history of the signal.
- A discrete signal can be represented by:

1. Graphical representation



2. Sequence representation

$$x(n) = \{\dots 0.9, -0.8, 0.7, 1.5, \underline{2}, 1.7, 1, 0.7, -0.8, 1.2 \dots\}$$

$$x(n) = \left\{\dots, 0, \underset{\uparrow}{2}, 1, 4, 1, 0, 0, \dots\right\}$$

- The bar under 2 (or an upward arrow ↑) indicate that 2 is the center of the origin point, where n=0. And (...) denote infinite elements on that side.

3. Functional representation, such as

$$x(n) = \begin{cases} 1, & for\ n = 1,3 \\ 4, & for\ n = 2 \\ 0, & elsewhere \end{cases}$$

4. Tabular representation, such as

n	...	-2	-1	0	1	2	3	4	5	...
x(n)	...	0	0	0	1	4	1	0	0	...

Classification of Signals

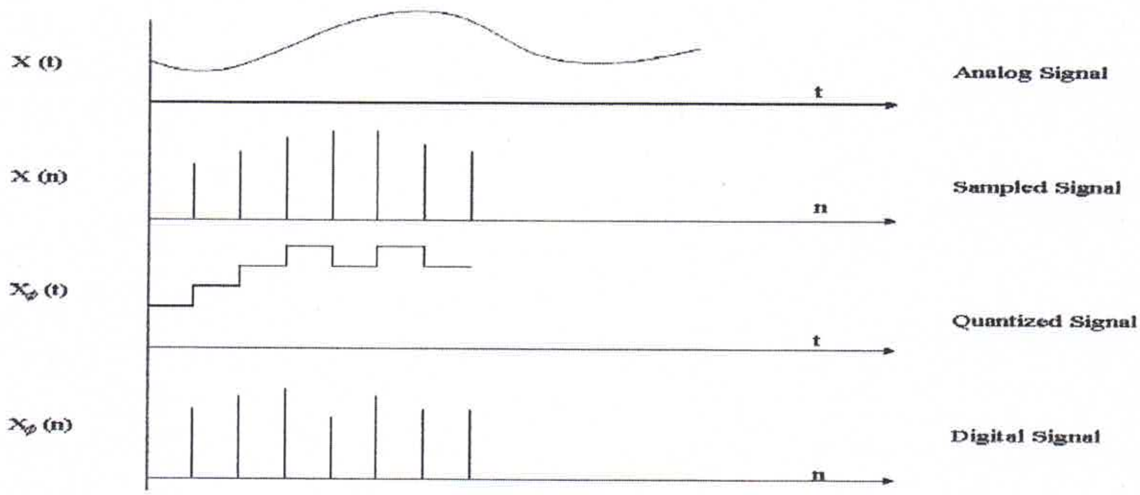
- Usually, signals are given in terms of plot in time domain. The signals can be categorized according to different parameters as follows;

1. **Based on Variables:**
   a. *f(t)=5t : single variable*
   b. *f(x,y)=2x+3y : two variables*
   c. *S_1= A Sin(wt) : real valued signal*
   d. *S_2 = A e^{jwt} : A Cos(wt)+j A Sin(wt) : Complex valued signal*
   e. $S_4(t)=\begin{bmatrix} S_1(t) \\ S_2(t) \\ S_3(t) \end{bmatrix}$: Multichannel signal

- Example: the color TV picture is a three-channel, three-dimensional signal, which can be represented by the vector

$$I(x,y,t) = \begin{bmatrix} I_r(x,y,t) \\ I_g(x,y,t) \\ I_b(x,y,t) \end{bmatrix} : \text{Multidimensional}$$

2. **Based on Representation:**

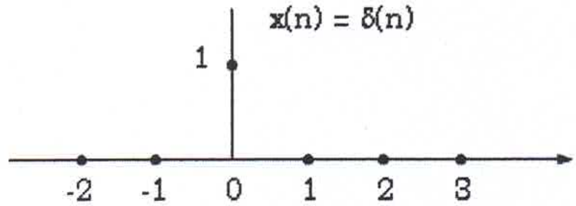


3. **Based on duration.**
   a. *Right sided: x(n)=0 for n<N*
   b. *Left sided: x(n)=0 for n>N*
   c. *Causal: x(n)=0 for n<0*
   d. *Anti-causal: x(n)=0 for n ≥0*
   e. *Non causal: x(n)=0 for |n|>N*

4. Based on the Shape.

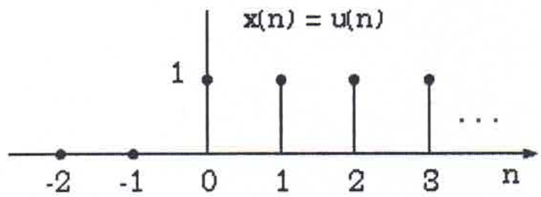
A. Unit Impulse:

$$\delta(n) = \begin{cases} 0, & elsewhere \\ 1, & n = 0 \end{cases}$$



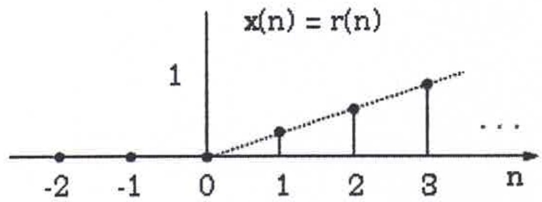
B. Unit Step:

$$u(n) = \begin{cases} 1, & n \geq 0 \\ 0, & n < 0 \end{cases}$$



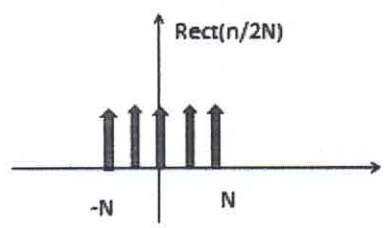
C. Unit Ramp:

$$r(n) = \begin{cases} n, & n \geq 0 \\ 0, & n < 0 \end{cases}$$



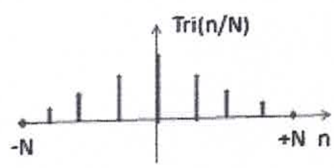
D. Rectangle signal

$$rect\left(\frac{n}{2N}\right) = \begin{cases} 1 \;...if\; |n| \leq N \\ 0 \;.........else \end{cases}$$



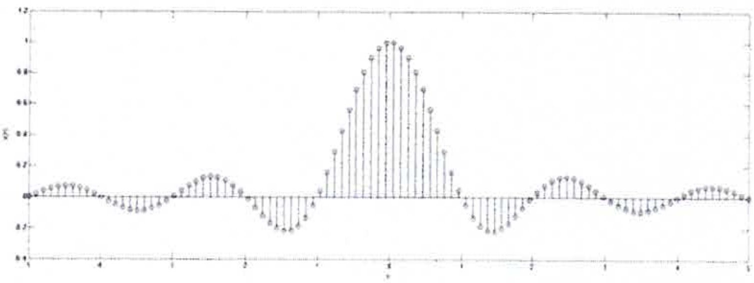
E. Triangle signal

$$tri\left(\frac{n}{N}\right) = \begin{cases} 1 - \frac{|n|}{N} \;... \;if\; |n| \leq N \\ 0 \;.........elsewhere \end{cases}$$



F. Discrete Sinc:

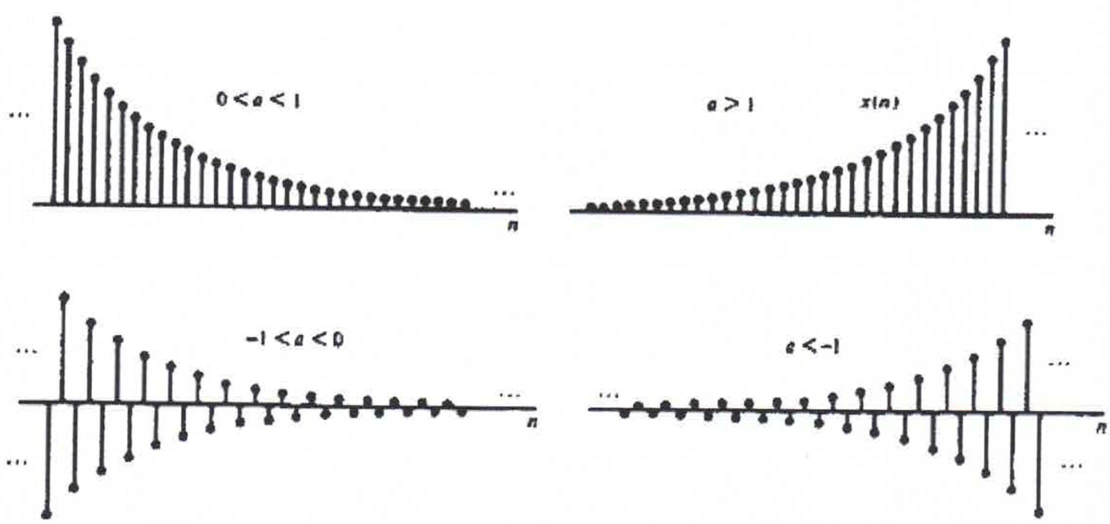
$$Sinc\left(\frac{n}{N}\right) = \frac{sin\left(\frac{n\pi}{N}\right)}{\left(\frac{n\pi}{N}\right)}$$



G. Exponential Sequence

$$x(n) = a^n$$

- If a real number, then x(n) is real.
- If 0<a<1, then sequence values are +ve and decreases with increasing n.
- For -1<a<0, the sequence values alternate in sign but again decreases in magnitude with increasing n.
- If a>1, then the sequences grows in magnitude as n increases.
- If -1>a, the sequence values alternate in sign but increase in magnitude with increasing n.



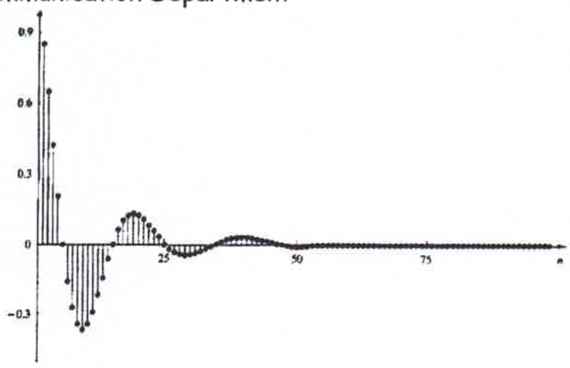
- When the parameter a is complex value, it can be expressed as

$$a \equiv re^{j\theta}$$

- Where r and θ are now the parameters. Hence, we can express x(n) as

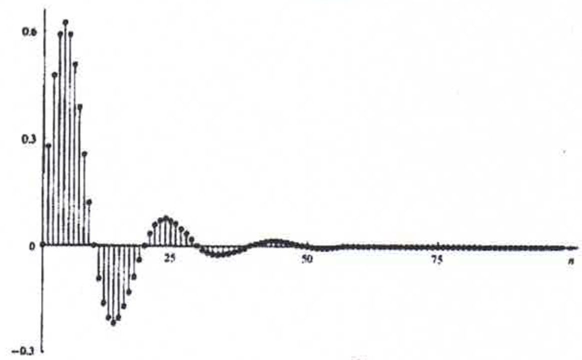
$$x(n) = r^n\, e^{j\theta n} = r^n(\cos\theta n + j\sin\theta n)$$

- If r>1, the sequence oscillates with exponentially growing envelope.
- If r<1, the sequence oscillates with exponentially decreasing envelope.



(a) Graph of $x_R(n) = (0.9)^n \cos\frac{\pi n}{10}$

Real

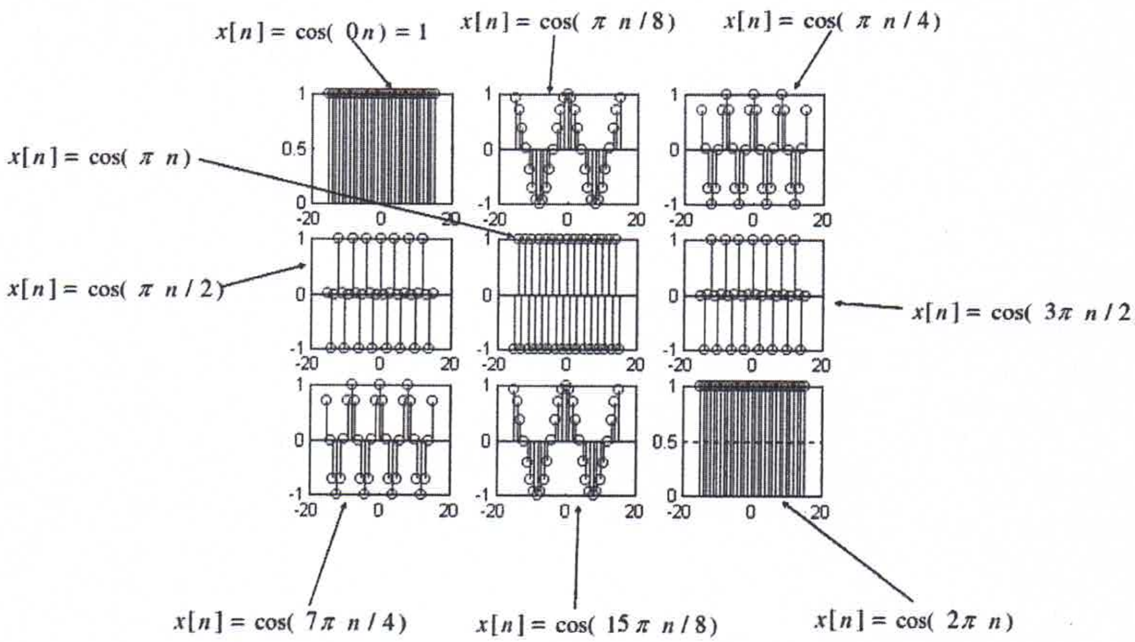


(b) Graph of $x_I(n) = (0.9)^n \sin\frac{\pi n}{10}$

imag

H. Sinusoidal Sequence

$$x(n) = A\cos(w_0 n + \theta)\ for\ all\ n$$



5. Energy & Power Signals

- The energy E of a signal *x(n)* is defined as

$$E \equiv \sum_{n=-\infty}^{\infty} |x(n)|^2$$

- If E is finite (i.e., $0 < E < \infty$), then *x(n)* is called an energy signal
- The signal energy of *x(n)* over the finite interval $-N < n < N$ as

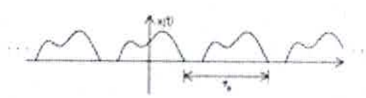
$$E = \sum_{n=-N}^{N} |x(n)|^2$$

- If x(n) is a periodic signal with fundamental period N and takes on finite values. Its power is given by

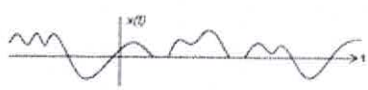
$$p = \frac{1}{N}\sum_{n=0}^{N-1}|x(n)|^2$$

- If P is finite (and nonzero), the signal is called a *power signal.*
- If *E* is finite, then P = 0.

6. **Periodic & non periodic based on repetition.**
   - The period of discrete signal is measured as the number of samples per period.
     x[n]=x[n± kN] k=0,1,2,3,4,....
   - The period N is the smallest number of samples that repeats., N: always an Integer
   - For combination of two or more signals N is the LCM (Least Common Multiple) of individual periods.



A Periodic signal with period *To*



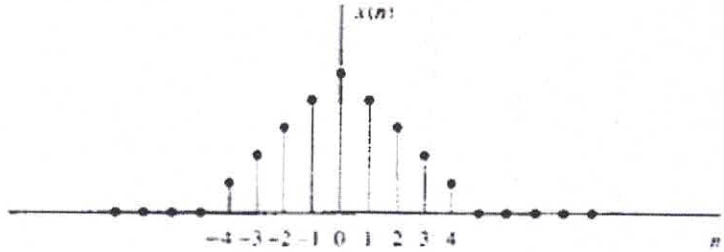
A nonperiodic signal

Example: the discrete sinusoidal signal $x(n) = A\sin 2\pi f_0 n$ is periodic when f_o is a rational number *(f_o=k/N)*

7. **Based on Symmetry**

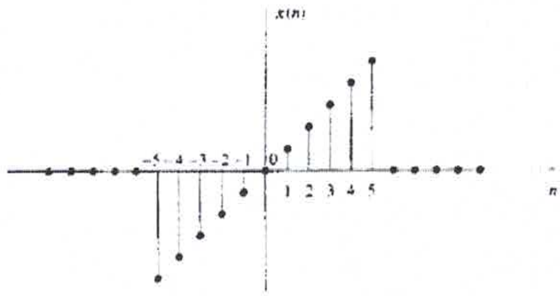
A. Symmetric (Even) ; The even signal component is formed by

$$x(-n) = x(n);\ x_e(n) = \frac{1}{2}[x(n) + x(-n)]$$



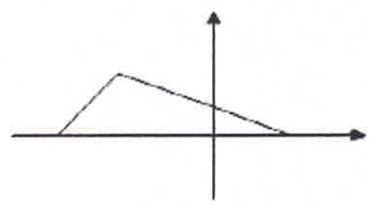
B. Anti-symmetric (Odd); The odd signal component is formed by

$$x(-n) = -x(n);\ x_o(n) = \frac{1}{2}[x(n) - x(-n)]$$

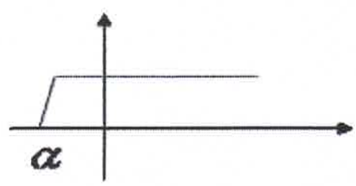


8. Signal Classification by duration & Area.

A. Finite duration: time limited.



B. Semi-infinite extent: right sided, if they are zero for t <a where a= finite



C. Left sided: zero for t >a