# What is Digital Signal Processing?

To understand what is Digital Signal Processing (DSP) let's examine what does each of its words mean.

Signal is any physical quantity that carries information. Processing is a series of steps or operations to achieve a particular end. It is easy to see that Signal Processing is used everywhere to extract information from signals or to convert information-carrying signals from one form to another. For example, our brain and ears take input speech signals, and then process and convert them into meaningful words. Finally, the word Digital in Digital Signal Processing means that the process is done by computers, microprocessor, or logic circuits.

The field DSP has expanded significantly over that last few decades as a result of rapid developments in computer technology and integrated-circuit fabrication. Consequently, DSP has played an increasingly important role in a wide range of disciplines in science and technology. Research and development in DSP are driving advancements in many high-tech areas including telecommunications, multimedia, medical and scientific imaging, and human-computer interaction.

### Concepts in Digital Signal Processing

The two main characters in DSP are signals and systems. A signal is defined as any physical quantity that varies with one or more independent variables such as time (onedimensional signal), or space (2-D or 3-D signal). Signals exist in several types. In the realworld, most of signals are continuous-time (analog signals) those have values continuously at every value of time. To be processed by a computer, a continuous-time signal has to be first sampled in time into a discrete-time signal so that its values at a discrete set of time instants can be stored in computer memory locations. Furthermore, in order to be processed by logic circuits, these signal values have to be quantized in to a set of discrete values, and the final coded result is called a digital signal. The terms discrete-time signal and digital signal can be used interchangeability to define two different formats (Fig. 1).

In signal processing, a system is defined as a process coder whose input and output are signals (Fig. 2).

### Signals Represent Information

Whether analog or digital, information is represented by the fundamental quantity in electrical engineering: the signal. Stated in mathematical terms, a signal is merely a function. Analog signals are continuous-valued; digital signals are discrete-valued. The independent variable of the signal could be time (speech), space (images), or the integers (denoting the sequencing of letters and numbers in the football score).









# **Sampling**

<u>Why sample?</u> Sampling is the necessary fundament for all digital signal processing and communication. Sampling can be defined as the process of measuring an analog signal at distinct points. Digital representation of analog signals offers advantages in terms of

- 1-Robustness towards noise, meaning we can send more bits/s.
- 2-Use of flexible processing equipment, in particular the computer.
- 3-More reliable processing equipment.
- 4-Easier to adapt complex algorithms.

Claude Shannon has been called the father of information theory, mainly due to his landmark papers on the "Mathematical theory of communication". Harry Nyquist was the first to state the sampling theorem in 1928, but it was not proven until Shannon proved it 21 years later in the paper "Communications in the presence of noise".

The following notations will be used: Original analog signal x(t), Sampling frequency  $f_s$ , Sampling interval  $T_s$  (Note that:  $f_s = 1/T_s$ ), Sampled signal  $x_s(n)$ . (Note that  $x_s(n) = x(nT_s)$ , Analogue angular frequency  $\Omega$ , and Digital angular frequency  $\omega$  (Note that:  $\omega = \Omega T_s$ ).

#### The Sampling Theorem

[[When sampling an analog signal the sampling frequency must be greater than twice the highest frequency component of the analog signal to be able to reconstruct the original signal from the sampled version]].

#### The process of sampling

We start with an analog signal. This can for example be the sound coming from your stereo at home or your friend talking. The signal is then sampled uniformly. Uniform sampling implies that we sample every  $T_s$  seconds. In Fig. 3, we see an analog signal. The analog signal has been sampled at times  $t = nT_s$ .



Fig. 3

In signal processing it is often more convenient and easier to work in the frequency domain. So let's look at the signal in frequency domain, Fig. 4. For illustration purposes we take the frequency content of the signal as a triangle. (If you Fourier transform the signal in Fig. 3 you will not get such a nice triangle.)



From Fig. 5, and according to the sample theorem, an aliasing-free condition appears. So, we are able to reconstruct the original signal exactly. How can we do this? will be explored further down under reconstruction. But first we will take a look at what happens when we sample too slowly.

#### Sampling too slowly

We will get overlap between the repeated spectra, see Fig. 6. The resulting spectra is the sum of these. This overlap gives rise to the concept of aliasing.



Fig. 6

To avoid aliasing we have to sample fast enough. But if we can't sample fast enough (possibly due to costs) we can include an Anti-Aliasing filter. This will not able us to get an exact reconstruction but can still be a good solution.

*Note:* Typically a low-pass filter that is applied before sampling to ensure that no components with frequencies greater than half the sample frequency remain.

#### **Reconstruction**

We want to recover the original signal, but the question is how?

The Answer: By using a simple reconstruction process. To achieve this we have to remove all the extra components generated in the sampling process. To remove the extra components we apply an ideal analog low-pass filter as shown in Fig. 7. As we see the ideal filter is rectangular in the frequency domain. A rectangle in the frequency domain corresponds to a sinc function in time domain (and vice versa).



Fig. 7



### **Discrete-Time Signals**

 A Discrete-time signal x(n) is a function of an independent integer variable n. The signal x(n) is not defined for noninteger values of n.

We can represent a discrete-time signal in different ways;

1. Graphical representation

Such as













### **Discrete-Time Signals and Systems**

Addition, Multiplication, and Scaling of Sequences

Amplitude Scaling: (A Constant Multiplier)



Addition of two signals (An Adder)

 $y(n) = x_1(n) + x_2(n), -\infty < n < \infty$ 





# Input-Output Description of Systems







### Input-Output Description of Systems

Solution (cont)

e

$$x(n) = \sum_{k=-\infty}^{n} x(k) = x(n) + x(n-1) + x(n-2) + \dots$$
  
This system is called an accumulator  
$$(0) = \sum_{k=-\infty}^{n} x(k) = x(0) + x(-1) + x(-2) + x(-3)$$

$$y(1) = \sum_{k=-\infty}^{1} x(k) = x(1) + x(0) + x(-1) + x(-2) + x(-3)$$
$$= 1 + 0 + 1 + 2 + 3 = 7$$
$$y(n) = [..., 0, 3, 5, 6, 6, 7, 9, 12, 12, ...]$$

.......

**Classification of Discrete-Time Systems** 

Linear System:

A system is linear if and only if

$$T[a_1x_1(n) + a_2x_2(n)] = a_1T[x_1(n)] + a_2T[x_2(n)]$$
  
=  $a_1y_1(n) + a_2y_2(n)$ 

for any arbitrary  $x_1(n)$  and  $x_2(n)$ , and any arbitrary constant  $a_1$  and  $a_2$ . It satisfies the superposition principle.



If the input is  $x_1(n)$ , the output will be  $y_1(n)$ if the input is  $x_2(n)$ , the output will be  $y_2(n)$ 

if the input is  $[x_1(n) + x_2(n)]$ , the output will be y(n)

if  $y(n) = [y_1(n) + y_2(n)]$ , then the system is linear.

Example 1:  $y(n) = e^{-x(n)}$ solution: for  $y_1(n) = e^{-x_1(n)}$ for  $y_2(n) = e^{-x_2(n)}$ for  $[x_1(n) + x_2(n)], \quad y(n) = e^{-[x_1(n) + x_2(n)]}$   $[y_1(n) + y_2(n)] = e^{-x_1(n)} + e^{-x_2(n)}$  $\neq e^{-[x_1(n) + x_2(n)]}$  It is non - linear system

### **Classification of Discrete-Time Systems**

Time-invariant Systems:

A system is called time-invariant if its input-output characteristics do not change with time.

*a*-shift the input  $x(n-n_0)$ 

b-shift the output  $y(n-n_0)$ 

if  $y(n-n_0) = y(n)$  then the system is time-invariant.

*Example 2* : *y*(n)=n *x*(n -1)

*solution*: Shift the input,  $y(n)=n x(n-n_0-1)$ 

shift the output,  $y(n-n_0)=(n-n_0) x(n-n_0-1)$ 

since  $y(n-n_0) \neq y(n)$ , then the system is time-variant.

Note: If the system ( time-invariant & linear ), it is called Linear Time Invariant Systems (LTI)

LTI system can easily be characterized by its output response  $\dot{h}(n)$  to the input  $\delta(n)$ . The input-output relationship can then be given by CONVOLUTION

$$y(n) = x(n) * h(n) = \sum_{k=-\infty}^{\infty} x(k)h(n-k)$$



<u>Example</u> 3: y(n) = x(n+1)At n = 0, y(0) = x(1), then the system is non-causal (anti-causal). <u>Example</u> 4:  $h(n) = 0.5^n u(n)$ Since h(n) = 0, for n < 0, then it is a causal system

# Classification of Discrete-Time Systems

bounded Sequence:

A Sequence x(n) is called bounded if  $|x(n)| \leq M_x \leq 00$ 

Stable Systems:

Difference Equation (D.E.) Representation

An arbitrary relaxed system is said to be bounded input-bounded output (BIBO) stable if and only if every bounded input produces a bounded output.

 $|x(n)| \le M_x < \infty \qquad |y(n)| \le M_y < \infty$ 

for all n. M, and M, are some finite numbers

An LTI System with h(n) is (BIBO) stable if  $\sum |h(n)| = M < \infty$ 

If the system does not satisfy any of those definitions, it is called unstable.

<u>Example</u> 5: y(n) = n x(n-1)As  $n \to \infty$ ,  $y(n) \to \infty$ , then the system is unstable. <u>Example</u> 6:  $h(n) = (1/4)^n u(n)$ Since  $\sum_{k=0}^{\infty} |h(n)| < \infty$ , it is a stable system

### Interconnection of Discrete-Time Systems

Discrete-time systems can be interconnected to form larger systems. They can be interconnected in serial or parallel.

In serial interconnection

$$\begin{array}{c} x(n) & \hline T_1[.] & y_1(n) & \hline T_2[.] & y_1(n) \\ \hline y_1(n) = \overline{T_1}[x(n)] & y(n) = \overline{T_2}[y_1(n)] \\ = \overline{T_2}[\overline{T_1}[x(n)]] \end{array}$$

If we combine  $T_1$  and  $T_2$  to , then  $y(n) = T_e[x(n)]$ 

If the systems  $T_1$  and  $T_2$  are linear and time invariant  $T_1 T_2 = T_2 T_1$ 

otherwise  $T_1 T_2 \neq T_2 T_1$ 



# **Convolution**



The process of computing the convolution between x(n) and h(k) involves the following steps.

- 1. Folding. Fold h(k) about k=0 to obtain h(-k)
- 2. Shifting. Shift h(-k) by  $n_0$  to the right (left) if  $n_0$  is positive (negative), to obtain  $h(n_0-k)$ .
- 3. *Multiplication*. Multiply x(k) by  $h(n_0-k)$  to obtain the product sequence
- 4. Summation. Sum all the values of the product sequence to obtain the value of the output at time  $n=n_0$
- 5. <u>Step 2 through 4 must be repeated, for all possible time shifts</u>.  $-\infty < n < \infty$

### Methods of Convolution

- 1- Graphical method.
- 2- Table-look up method.
- 3- Vector-by-matrix method.
- 4- Add-overlap method.
- 5- Analytical method.

# 1- Graphical Method











# 2- Table-look up Method h(n)

$$x(n) = \begin{bmatrix} 1 & 2 & 3 & 1 \end{bmatrix}$$
,  $h(n) = \begin{bmatrix} 1 & 2 & 1 & -1 \end{bmatrix}$ 

find 
$$y(n)$$
?



$$y(n) = \begin{bmatrix} 1 & 4 & 8 & 8 & 3 & -2 & -1 \end{bmatrix}$$

# 3- Vector-by-matrix Method

$$x(n) = \begin{bmatrix} 1 & 2 & 3 & 1 \end{bmatrix}$$
,  $h(n) = \begin{bmatrix} 1 & 2 & 1 & -1 \end{bmatrix}$ 

find y(n)?

$$\begin{bmatrix} 1 & 0 & 0 & 0 \\ 2 & 1 & 0 & 0 \\ 3 & 2 & 1 & 0 \\ 1 & 3 & 2 & 1 \\ 0 & 1 & 3 & 2 \\ 0 & 0 & 1 & 3 \\ 0 & 0 & 0 & 1 \end{bmatrix} \begin{bmatrix} 1 \\ 2 \\ 1 \\ -1 \end{bmatrix} = \begin{bmatrix} 1 \\ 4 \\ 8 \\ 8 \\ 3 \\ -2 \\ -1 \end{bmatrix}$$

# 4- Add-overlap Method

$$x(n) = [1 \ 2 \ 1 ]$$
,  $h(n) = [1 \ -1 \ 2 \ 1 \ 2 \ -1 \ 1 \ 3 \ 1 ]$ 

	1	2	1
1	1	2	1
-1	-1	-2	-1
2	2	4	2

 $y_1(n) = \begin{bmatrix} 1 & 1 & 1 & 3 & 2 \end{bmatrix}$ 

	1	2	1
1	1	2	1
2	2	4	2
-1	-1	-2	-1

 $y_2(n) = \begin{bmatrix} 1 & 4 & 4 & 0 & -1 \end{bmatrix}$ 

	1	2	1
1	1	2	1
3	3	6	3
1	1	2	1

 $y_3(n) = \begin{bmatrix} 1 & 5 & 8 & 5 & 1 \end{bmatrix}$ 

$$y_1(n) = \begin{bmatrix} 1 & 1 & 1 & 3 & 2 \end{bmatrix}$$
Shifted  $y_2(n) = \begin{bmatrix} 0 & 0 & 0 & 1 & 4 & 4 & 0 & -1 \end{bmatrix}$ 
Shifted  $y_3(n) = \begin{bmatrix} 0 & 0 & 0 & 0 & 0 & 0 & 1 & 5 & 8 & 5 & 1 \end{bmatrix}$ 

$$y(n) = y_1(n) + Shifted y_2(n) + Shifted y_3(n)$$
So,  $y(n) = \begin{bmatrix} 1 & 1 & 1 & 4 & 6 & 4 & 1 & 4 & 8 & 5 & 1 \end{bmatrix}$ 

# 5- Analytical Method

$$y(n) = \sum_{k=-\infty}^{\infty} x(k) h(n-k)$$
postion of  $N_1 \le x(n) \le N_2$ 
postion of  $M_1 \le h(n) \le M_2$ 

$$y(n) = \sum_{k=k_l}^{k_u} x(k) h(n-k)$$
How to calculate  $k_u$  and  $k_l$ ?
for  $x(n)$ ,  $N_1 \le k \le N_2$ 
for  $h(n)$ ,  $M_1 \le n-k \le M_2$ 
or  $M_1 - n \le -k \le M_2 - n$ 
i.e,  $n - M_2 \le k \le n - M_1$ 
 $\therefore k_l = max \{N_1, n - M_2\}$ 
and  $k_u = min\{N_2, n - M_1\}$ 

Example: Find y(n) if x(n) = u(n) - u(n-10) and  $h(n) = \left(\frac{1}{2}\right)^n u(n)$ 

Solution:-

$$\begin{split} N_{1}=0 , & N_{2}=9 \\ M_{1}=0 & M_{2}=\infty \\ k_{l} &= max\{N_{1}, n-M_{2}\} = max\{0, n-\infty\} = 0 \\ k_{u} &= min\{N_{2}, n-M_{1}\} = min\{9, n-0\} = min\{9, n\} = \begin{cases} n \ for \ n \leq 9 \\ 9 \ for \ n > 9 \end{cases} \end{split}$$

for  $0 \le n \le 9$ 

$$y(n) = \sum_{k=0}^{n} x(k) h(n-k) = \sum_{k=0}^{n} \left(\frac{1}{2}\right)^{n-k} = \left(\frac{1}{2}\right)^{n} \sum_{k=0}^{n} \left(\frac{1}{2}\right)^{-k} = \left(\frac{1}{2}\right)^{n} \sum_{k=0}^{n} \left[\left(\frac{1}{2}\right)^{-1}\right]^{k} = \left(\frac{1}{2}\right)^{n} \left[\frac{1-\left[\left(\frac{1}{2}\right)^{-1}\right]^{n+1}}{1-\left(\frac{1}{2}\right)^{-1}}\right] = \left(\frac{1}{2}\right)^{n} \left[\frac{1-\left[\left(\frac{1}{2}\right)\right]^{n}}{-1}\right]$$

The last step above is obtained by using the geometrical progression formula given in appendix B ( page 317 in (fundamentals of digital signal processing book) by

$$\sum_{k=0}^{n} (b)^{k} = \left[\frac{1-(b)^{n+1}}{1-b}\right] \quad \text{for } b \neq 1$$

$$for n > 9$$

$$y(n) = \sum_{k=0}^{9} x(k) h(n-k) = \sum_{k=0}^{9} \left(\frac{1}{2}\right)^{n-k} = \left(\frac{1}{2}\right)^{n} \left[1 + \left(\frac{1}{2}\right)^{-1} + \left(\frac{1}{2}\right)^{-2} + \dots + \left(\frac{1}{2}\right)^{-9}\right]$$
H.W
Find y(n) of the following system:



# **De-convolution**

# 1-Iterative method

Example: 
$$h(n) = [4 \ 2 \ 3 - 4]$$
,  $y(n) = [4 \ 14 \ 17 \ 9 \ -6 \ -8]$   
find  $x(n)$ ?  
 $y(n) = \sum_{k=0}^{n} x(k) h(n-k)$   
 $y(0) = \sum_{k=0}^{0} x(k) h(-k) = x(0) \cdot h(0)$   
 $x(0) = \frac{y(0)}{h(0)} = \frac{4}{4} = 1$   
 $y(1) = \sum_{k=0}^{1} x(k) h(1-k) = x(0) \cdot h(1) + x(1) \cdot h(0)$   
 $x(1) = \frac{y(1) - x(0) \cdot h(1)}{h(0)} = \frac{14 - (1 \times 2)}{4} = 3$   
 $y(2) = \sum_{k=0}^{2} x(k) h(2-k) = x(0) \cdot h(2) + x(1) \cdot h(1) + x(2) \cdot h(0)$   
 $x(2) = \frac{y(2) - x(0) \cdot h(2) - x(1) \cdot h(1)}{h(0)} = \frac{17 - (1 \times 3) - (3 \times 2)}{4} = 2$   
 $y(3) = \sum_{k=0}^{3} x(k) h(3-k) = x(0) \cdot h(3) + x(1) \cdot h(2) + x(2) \cdot h(1) + x(3) \cdot h(0)$   
 $x(3) = \frac{y(3) - x(0) \cdot h(3) - x(1) \cdot h(2) - x(2) \cdot h(1)}{h(0)} = 0$ 

For y(n) = x(n) \* h(n), if length of x(n) = N, and length of h(n) = Mthen length of y(n) = (N + M) - 1

For the above example, N = 3, and M = 4, then (N + M) - 1 = 6,

# 2-The Polynomial Method

Example:  $h(n) = \begin{bmatrix} 4 & 2 & 3 & -4 \end{bmatrix} \Rightarrow B(p) = 4 + 2p + 3p^2 - 4p^3$  $y(n) = \begin{bmatrix} 4 & 14 & 17 & 9 & -6 & -8 \end{bmatrix} \Rightarrow C(p) = 4 + 14p + 17p^2 + 9p^3 - 6p^4 - 8p^5$ 

Since 
$$y(n) = x(n) * h(n)$$
, then  $C(p) = A(p) \cdot B(p)$  or  $A(p) = \frac{C(p)}{B(p)}$ 

$$4 + 2p + 3p^2 - 4p^3$$

$$4 + 14p + 17p^2 + 9p^3 - 6p^4 - 8p^5$$

# 3-The Graphical Method

For the same example:  $h(n) = \begin{bmatrix} 4 & 2 & 3 & -4 \end{bmatrix}$  and  $y(n) = \begin{bmatrix} 4 & 14 & 17 & 9 & -6 & -8 \end{bmatrix}$ 

STEP ONE



#### STEP TWO



#### STEP THREE



If we calculate x(3) and so on, they will be zero. Why?

### Linear Constant-Coefficient Difference Equations (LCCDEs)

Remembering linear differential equations

$$\frac{dy(t)}{dt} - y(t) = x(t)$$

A difference equation is the discrete-time analogue of a differential equation. We simply use differences [ x (n) - x (n-1)] rather than derivatives (  $\frac{dx(t)}{dt}$ ).

An important subclass of linear systems are those whose input is x(n), output is y(n), and satisfying the following  $N^{th}$  - order LCCDE:

$$\sum_{k=0}^{N} a_k y(n-k) = \sum_{r=0}^{M} b_r x(n-r) \qquad a_0 \neq 0$$

If the system is causal, then we can rearrange the above Eq. as

$$y(n) = -\sum_{k=1}^{N} \frac{a_k}{a_0} y(n-k) + \sum_{k=0}^{M} \frac{b_r}{a_0} x(n-r)$$

### Solutions of Linear Constant- Coefficient Difference Equations

#### First -order LCCDE

Example -1: Solve the following DE for y(n), assuming y(n) = 0 for all n < 0and  $x(n) = \delta(n)$ .

$$y(n) - ay(n-1) = x(n)$$

This corresponds to calculating the response of the system when excited by an impulse, assuming "zero initial conditions"

Solution: Rewrite	y(n) = ay(n-1) + x(n)	
Evaluate:	y(0) = ay(-1) + x(0) =	1
	y(1) = ay(0) + x(1) =	а
	y(2) = ay(1) + x(2) =	a <sup>2</sup>

For all n>0, It can be written that

$$y(n) = a^n$$

Since the response of the system for n < 0 is defined to be zero, the unit sample response becomes  $h(n) = a^n u(n)$ 

If |a| < 1, then the system is .....? If |a| > 1, then the system is .....?

N<sup>th</sup> -order LCCDE

Two methods Direct method Indirect Method (z-transform)

**Direct solution Method:** 

The total solution is the sum of two parts

- Part 1 homogeneous solution
- Part 2 particular solution

The Homogeneous solution

Assuming that the input . Since , this gives us the <u>zero-input response</u> of the system

$$\sum_{k=0}^{N} a_k y(n-k) = 0$$

The solution is the form of an exponential

$$y_h(n) = \lambda^n$$

substitute this in the previous equation.

$$\lambda^{n} + a_{1}\lambda^{n-1} + a_{2}\lambda^{n-2} + \dots + a_{N-1}\lambda^{n-(N-1)} + a_{N}\lambda^{n-N} = 0$$

$$\lambda^{n-N} \left( \lambda^{N} + a_1 \lambda^{N-1} + a_2 \lambda^{N-2} + \dots + a_{N-1} \lambda + a_N \right) = 0$$

This is called characteristic polynomial of the system. It has N roots and denotes by  $\lambda_1, \lambda_2, ..., \lambda_N$ 

The roots can be real or complex or some roots are identical.

Let assume that roots are real and not identical, the solution becomes

$$y_h(n) = C_1 \lambda_1^n + C_2 \lambda_2^n + \dots + C_N \lambda_N^n$$

The coefficients  $C_i$ , i = 1, 2, ..., N are determined from the initial conditions.

If there are two identical roots, the solution becomes

$$y_h(n) = C_1 \lambda_1^n + C_1 n \lambda_1^n + C_2 \lambda_2^n + \dots + C_N \lambda_N^n$$

Example-2:

Find the zero-input response for the second-order difference equation

$$y(n) - 3y(n-1) - 4y(n-2) = 0$$

 $\lambda_1=-1,\,\lambda_2=4$ 

The homogeneous solution form  $y_h(n) = \lambda^n$ 

$$\lambda^{n} - 3\lambda^{n-1} - 4\lambda^{n-2} = 0$$
$$\lambda^{n-2} (\lambda^{2} - 3\lambda - 4) = 0$$

The homogenous solution is

$$y_h(n) = C_1 \lambda_1^n + C_2 \lambda_2^n$$
  
=  $C_1 (-1)^n + C_2 4^n$ 





The total solution of the difference equation

$$y(n) = y_h(n) + y_p(n)$$

Example:-4

Determine the response  $y(n), n \ge 0$  of the system described by the second-order difference equation

$$y(n) = 0.7y(n-1) - 0.1y(n-2) + 2x(n) - x(n-2)$$

to the input  $x(n) = 4^n u(n)$ 

The homogenous solution is

$$y(n) - 0.7 y(n-1) + 0.1 y(n-2) = 0$$
  
$$\lambda^{n} - 0.7 \lambda^{n-1} + 0.1 \lambda^{n-2} = 0$$
  
$$\lambda^{n+2} (\lambda^{2} - 0.7 \lambda + 0.1) = 0 \Rightarrow \lambda_{1} = 0.5 \Rightarrow \text{and} \quad \lambda_{2} = 0.2$$
  
$$y_{h}(n) = c_{1} 0.5^{n} + c_{2} 0.2^{n}$$







### Frequency Response of LTI Systems

The Fourier representation of signals plays an extremely important role in both continuous-time and discrete-time signal processing. It provides a method for mapping signals into another "domain" in which to manipulate them. What makes the Fourier representation particularly useful is the property that the convolution operation is mapped to multiplication. In addition, the Fourier transform provides a different way to interpret signals and systems. In this section, we will develop the discrete-time Fourier transform (*i.e.*, a Fourier transform for discrete-time signals). We will show how complex exponentials of linear time-invariant (LTI) systems and how this property leads to the notion of a frequency response representation of LSI systems.

#### A. Response to Complex Exponential

Let  $x(n) = e^{j\omega n}$  be input into an LTI system with causal impulse response h(n). The output is

$$y(n) = h(n) * x(n) = h(n) * e^{j\omega n} = \sum_{k=-\infty}^{\infty} h(k) x(n-k)$$
$$= \sum_{k=-\infty}^{\infty} h(k) e^{j\omega(n-k)} = e^{j\omega n} \sum_{k=-\infty}^{\infty} h(k) e^{-j\omega k}$$

Let us define  $H(e^{j\omega})$ : a function of  $\omega$ , as  $\omega$  varies form  $(-\infty to + \infty)$  to be

$$H(e^{j\omega}) \triangleq \sum_{k=-\infty}^{\infty} h(k) e^{-j\omega k}$$
$$y(n) = e^{j\omega n^{INPUT}} H(e^{j\omega})^{frequency response}$$
$$y(n) = x(n) H(e^{j\omega})$$

or

•  $H(e^{j\omega})$  = frequency response function (a conjugate symmetric function of  $\omega$ )

$$H(e^{j\omega}) = H_{Re}(e^{j\omega}) + jH_{Im}(e^{j\omega}) = |H(e^{j\omega})| e^{j \arg H(e^{j\omega})}$$
$$= |H(e^{j\omega})| e^{j \varphi(e^{j\omega})}$$

•  $|H(e^{j\omega})|$  = Magnitude response (an even function of  $\omega$ ) •  $\arg H(e^{j\omega}) = \emptyset(e^{j\omega})$  = Phase response (an odd function of  $\omega$ ) •  $H(e^{j\omega})$  is periodic with period = 2  $\pi$ where the magnitude and phase of  $H(e^{j\omega})$  are given by  $|H(e^{j\omega})| = [H_{Re}^2(e^{j\omega}) + H_{Im}^2(e^{j\omega})]^{1/2}$ 

$$\arg H(e^{j\omega}) = \emptyset(e^{j\omega}) = \tan^{-1}[H_{Im}(e^{j\omega})/H_{Re}(e^{j\omega})]$$

#### B. Response to Sinusoidal

Now let 
$$x(n) = A \cos(\omega_o n + \theta) = \frac{Ae^{j\omega_o n} e^{j\theta}}{2} + \frac{Ae^{-j\omega_o n} e^{-j\theta}}{2}$$
 be input into an LTI system with causal impulse response  $h(n)$ .

Because of linearity the response can be found by adding the responses of the complex exponential sequences of  $\frac{Ae^{j\omega_0 n}e^{j\theta}}{2}$  and  $\frac{Ae^{-j\omega_0 n}e^{-j\theta}}{2}$  the output becomes

$$y(n) = A H(e^{j\omega_0 n}) e^{j\omega_0 n} e^{j\theta} / 2 + A H(e^{-j\omega_0 n}) e^{-j\omega_0 n} e^{-j\theta} / 2$$

The second part of y(n) is seen to be the complex conjugate of the first part, thus y(n) becomes two times the real part of either; that is

$$y(n) = 2 \operatorname{Re} \left[ \frac{A}{2} H(e^{j\omega_0 n}) e^{j\omega_0 n} e^{j\theta} \right]$$
$$y(n) = 2 \operatorname{Re} \left[ \frac{A}{2} |H(e^{j\omega_0})| e^{j\phi(e^{j\omega_0})} e^{j\omega_0 n} e^{j\theta} \right]$$
$$y(n) = A \operatorname{Re} \left\{ |H(e^{j\omega_0})| e^{[j(\omega_0 n + \theta + \phi(e^{j\omega_0}))]} \right\}$$
$$y(n) = A |H(e^{j\omega_0})| \cos(\omega_0 n + \theta + \phi(e^{j\omega_0}))$$
$$\underset{\text{Magnitude}}{\operatorname{Change in phase}}$$

Therefore, it has been shown that the output to sinusoid is another sinusoid of the same frequency but with different phase and different magnitude.

Example -1:- Find the frequency response of LTI system characterized by unit sample response (impulse response)

$$h(n) = a^n u(n) \quad for \ |a| \ < 1$$

Solution: It is an IIR system

By definition the frequency response  $H(e^{j\omega})$  is given by

$$H(e^{j\omega}) = \sum_{n=-\infty}^{\infty} h(n) e^{-j\omega n} = \sum_{n=0}^{\infty} a^n e^{-j\omega n}$$
$$= \sum_{n=0}^{\infty} \left(a e^{-j\omega}\right)^n = \frac{1}{1-a e^{-j\omega}} = \frac{1}{(1-a\cos\omega)+j(a\sin\omega)}$$

Mag. response=  $|H(e^{j\omega})| = \frac{1}{[(1-a\cos\omega)^2 + (a\sin\omega)^2]^{1/2}} = \frac{1}{(1+a^2-2a\cos\omega)^{1/2}}$ Phase response=  $\emptyset(e^{j\omega}) = -\tan^{-1}[a\sin\omega/(1-a\cos\omega)]$  Digital Signal Processing (DSP)

Plot Mag. and Phase responses for a = 0.5.

Example -2:- For an LTI discrete-time system with the following impulse response

 $h(n) = \delta(n-1) - 2 \delta(n-3) + \delta(n-5)$ 

(i) Give expressions for h(n) in terms of unit steps and then in vector form.

(*ii*) Plot such impulse response h(n).

(iii) Specify whether the system is of the FIR or of the IIR type, Why?

(*iv*) Find the frequency response  $H(e^{j\omega})$ .

(v) Find and plot magnitude and phase responses.

(vi) Compute the unit step response.

(*vii*)Compute the response to  $x(n) = 4 \cos[\frac{\pi}{4}(n-2)]$ . Then calculate the corresponding time delay of the system if the sampling rate is 8 k sample/sec.

Solution:

(i) Using the fact that a unit sample can be written as the difference of two steps as follows:

$$\delta(\mathbf{n}) = u(\mathbf{n}) - u(\mathbf{n} - 1)$$

Therefore, h(n) = [u(n-1) - u(n-2)] - 2[u(n-3) - u(n-4)] + [u(n-5) - u(n-6)]*i.e.*, h(n) = u(n-1) - u(n-2) - 2u(n-3) + 2u(n-4) + u(n-5) - u(n-6)In vector form,

 $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$ (ii) The plot of such impulse response h(n)is shown here.  $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$   $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$   $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$   $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$   $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$   $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$   $h(n) = \begin{bmatrix} 0 & 1 & 0 & -2 & 0 & 1 \end{bmatrix}$ 

(*iii*) The system is of the FIR type, because h(n) is of finite duration.

$$(iv) H(e^{j\omega}) = \sum_{-\infty}^{\infty} h(n) e^{-j\omega n} = e^{-j\omega} - 2 e^{-j3\omega} + e^{-j5\omega} = = e^{-j3\omega} [e^{j2\omega} - 2 + e^{-j2\omega}] = e^{-j3\omega} [-2 + 2\cos(2\omega)] = e^{-j3\omega} (-1) [2 - 2\cos(2\omega)] = e^{-j3\omega} (e^{-j\pi}) [2 - 2\cos(2\omega)] = e^{-j(3\omega + \pi)} [2 - 2\cos(2\omega)]$$

(v) From the above frequency response,

Magnitude response =  $|H(e^{j\omega})| = [2 - 2\cos(2\omega)]$ .

Phase response =  $\emptyset(e^{j\omega}) = -(3\omega + \pi) = -3\omega - \pi$ 



(vi) The unit step response is  $u(n) * h(n) = u(n) * [\delta(n-1) - 2\delta(n-3) + \delta(n-5)]$ 

$$= u(n-1) - 2 u(n-3) + u(n-5).$$
(vii) The response to  $x(n) = 4 \cos[\frac{\pi}{4}(n-2)]$ 

$$\omega_0 = \frac{\pi}{4}, \text{ So } |H(e^{j\omega_0})| = [2 - 2\cos(2\omega_0)] = [2 - 2\cos(\frac{\pi}{2})] = 2.$$

$$\emptyset(e^{j\omega_0}) = -3\omega_0 - \pi = -\frac{3\pi}{4} - \pi = -\frac{7\pi}{4}$$

$$y(n) = 4.(2) \cdot \cos[\frac{\pi}{4}n - \frac{\pi}{2} - \frac{7\pi}{4})] = 8 \cdot \cos[\frac{\pi}{4}(n-2-7)]$$

$$= 8 \cdot \cos[\frac{\pi}{4}(n-9)]$$
Delay = 9-2=7 samples, Time delay = 7  $T_s = 7/f_s = 7/8000 = 0.000875$  sec.  
= 0.875 m sec.

Example -3:- If the step response of an LTI system is

$$y_u(n) = [1 \ 3 \ 2 \ 2 \ 3 \ 1],$$

find the unit sample response h(n). Then find magnitude and phase responses. Is the system possesses a linear phase response? Plot it.

 $\begin{aligned} \underline{Solution:} \\ \text{It is known that } \delta(n) &= u(n) - u(n-1), \text{ So} \\ h(n) * \delta(n) &= h(n) * [u(n) - u(n-1)] \\ \text{or} & h(n) * \delta(n) &= h(n) * u(n) - h(n) * u(n-1), \\ i.e., & h(n) &= y_u(n) - y_u(n-1) = [1 \ 3 \ 2 \ 2 \ 3 \ 1 \ 0] - [0 \ 1 \ 3 \ 2 \ 2 \ 3 \ 1] \\ h(n) &= [1 \ 2 \ -1 \ 0 \ 1 \ -2 \ -1] \\ H(e^{j\omega}) &= 1 + 2 \ e^{-j\omega} - e^{-j2\omega} + 0 \ . \ e^{-j3\omega} + e^{-j4\omega} - 2 \ e^{-j5\omega} - e^{-j6\omega} \\ H(e^{j\omega}) &= e^{-j3\omega} \left[ (e^{j3\omega} - e^{-j3\omega}) + 2 (e^{j2\omega} - e^{-j2\omega}) - (e^{j\omega} - e^{-j\omega}) \right] \\ H(e^{j\omega}) &= e^{-j3\omega} \left[ 2j \sin(3\omega) + 4 j\sin(2\omega) - 2 j\sin(\omega) \right] \\ H(e^{j\omega}) &= e^{j\pi/2} \ e^{-j3\omega} \left[ 2 \sin(3\omega) + 4 \sin(2\omega) - 2 \sin(\omega) \right] \\ H(e^{j\omega}) &= e^{-j(3\omega - \frac{\pi}{2})} \left[ 2 \sin(3\omega) + 4 \sin(2\omega) - 2 \sin(\omega) \right] \end{aligned}$ 

Mag. Response =  $|H(e^{j\omega})| = 2\sin(3\omega) + 4\sin(2\omega) - 2\sin(\omega)$ Phase Response =  $\emptyset(e^{j\omega}) = -3\omega + \frac{\pi}{2}$ ; Yes linear phase, <u>plot it</u>.