

IT4440 /IT4489

FUNDAMENTAL OF DIGITAL SIGNAL

(MULTIMEDIA AND GAMES)

OUTLINE

- ☐ Introduction to DSP.
- ☐ Principles and operation of DSP.
- ☐ Advantages and disadvantages of DSP.
- ☐ Applications.
- ☐ Conclusion.



MULTIMEDIA DEFINITION

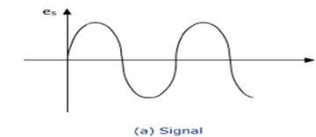
Multimedia is the field concerned with the computer controlled integration of text, graphics, drawings, image, video, animation, audio, and any other media where every type of information can be represented, stored, transmitted and processed digitally

Digitally ?

WHAT IS SIGNAL?

A physical phenomenon that carries or convey information from one place to other and represents as a function of independent variables such as time, distance, etc.

Anything that carries information can be called as signal



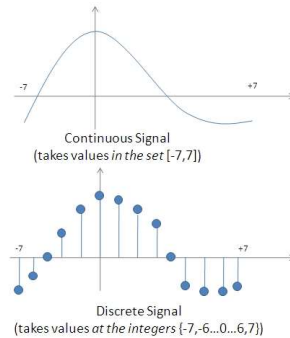
SOME DEFINITIONS

Continuous Signals: a continuum of time and are thus, represented by a continuous independent variable (often referred to as analog signals)

A signal $x(t)$ is said to be a continuous time signal if it is defined for all time t .

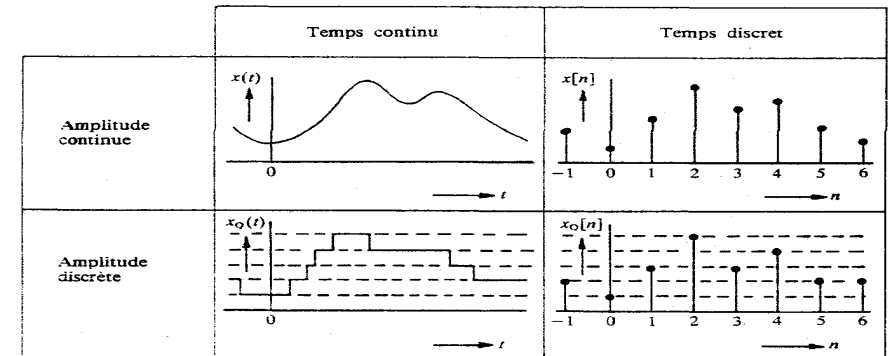
Discrete Signals: discrete times are known as discrete signals(sequence of numbers)

A discrete time signal $x[nT]$ has values specified only at discrete points in time



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SIGNAL

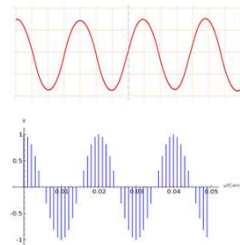


WHAT IS SIGNAL PROCESSING?

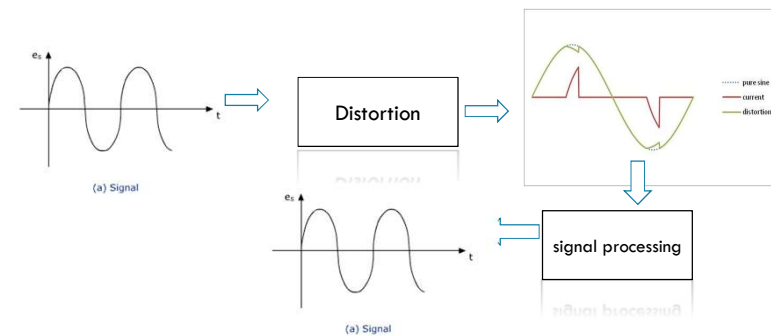
The process of operation in which the characteristics of a signal (Amplitude, shape, phase, frequency, etc.) undergoes a change is known as signal processing.

Types of signal processing:

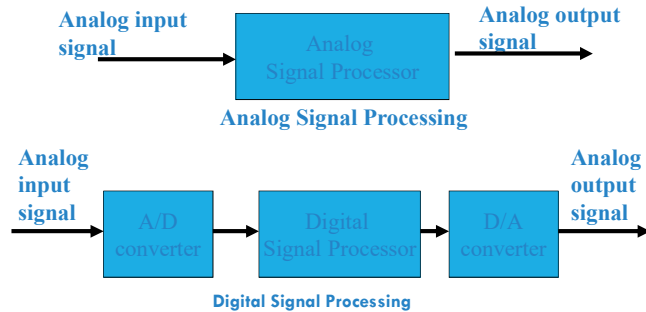
1. Analog signal processing
2. Digital signal processing



NEED OF SIGNAL PROCESSING ?



BASIC ELEMENTS OF A SIGNAL PROCESSING SYSTEM



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ADVANTAGES OF DIGITAL OVER ASP

- flexibility in reconfiguring the DSP operations simply.

(Reconfiguration of an analogue system usually implies a redesign of hardware, testing and verification that it operates properly)

- DSP provides better control of **accuracy** requirements.
- Digital signals are **easily stored** on magnetic media
- The DSP allows for the implementation of more **sophisticated** signal processing algorithms.

In some cases a digital implementation of the signal processing system is cheaper than its analogue counterpart.

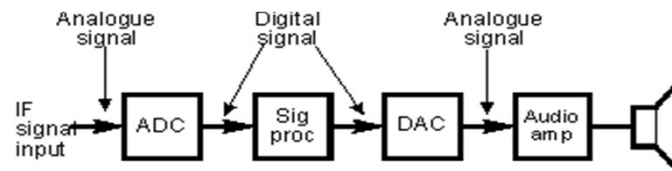
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DSP SYSTEM

DIGITAL: Operating by the use of discrete signal to represent data in the **form of numbers**.

SIGNAL: A parameter (electrical quantity or effect) that can be varied in such a way as to **convey information**.

PROCESSING: a **series operation** performed according to programmed instructions.



ADC



- ✓ Converting a continuously changing waveform (analog) into a series of discrete levels (digital)
- ✓ The analog waveform is sliced into equal segments and the waveform amplitude is measured in the middle of each segment
- ✓ The collection of measurements make up the digital representation of the waveform

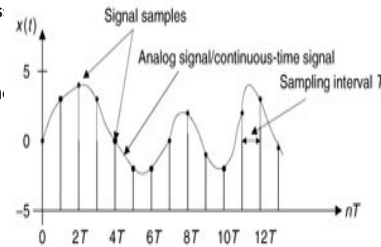
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ANALOG TO DIGITAL AND DIGITAL TO ANALOG CONVERSION

A/D conversion can be viewed as a three step process

1. **Sampling**: is the conversion of a continuous time signal into a discrete time signal obtained by taking "samples" of the continuous time signal at discrete time instants.

Thus, if $x(t)$ is the input to the sampler, the output is $x(nT)$, where T is called the *Sampling interval*.



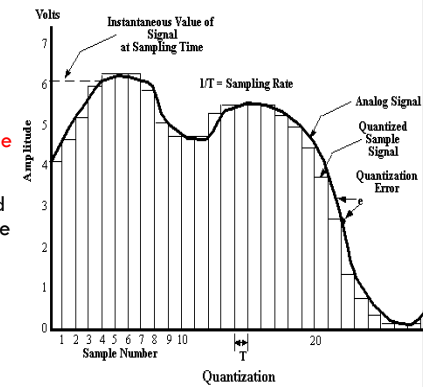
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2. QUANTIZATION

is the **conversion of discrete time continuous valued signal into a discrete-time discrete-value (digital) signal**.

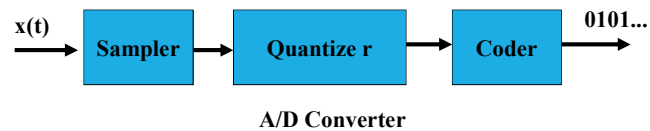
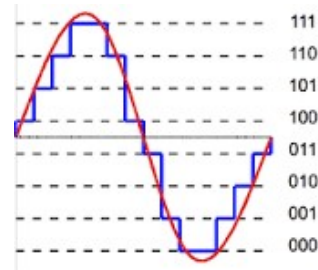
The value of each signal sample is represented by a value selected from a finite set of possible values.

The difference between unquantized sample and the quantized output is called the **Quantization error**.



STEP 3. CODING

In the coding process, each discrete value is represented by a b-bit binary sequence.



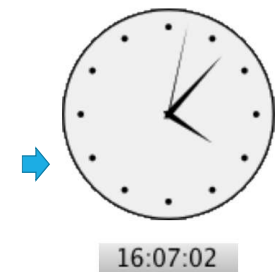
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EXAMPLE

ADC including 2 steps

- Sampling
- Quantization

Eg from analog signal to digital



16:07:02

HOW WE DO

A very important consideration in data communications is how fast we can send data, in bits per second, over a channel.

Data rate depends on three factors:

1. The bandwidth available
2. The level of the signals we use
3. The quality of the channel (the **level of noise**)
 - Noiseless Channel: Nyquist Bit Rate
 - Noisy Channel: Shannon Capacity Topics discussed in this section

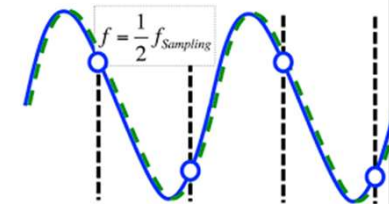
NYQUIST THEOREM

For lossless digitization, the sampling rate should be at least twice the maximum frequency response.

In mathematical terms:

$$f_s > 2 * f_m$$

where f_s is sampling frequency and f_m is the maximum frequency in the signal



NYQUIST THEOREM

- Nyquist gives the upper bound for the bit rate of a transmission system
- Nyquist theorem states that for a noiseless channel:

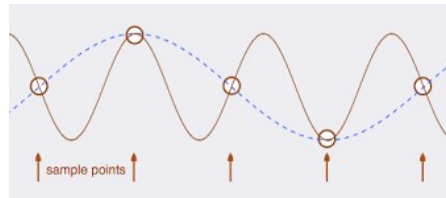
$$C = 2 B \log_2 2n$$

Where : C= capacity in bps

B = bandwidth in Hz

2n: = discrete levels

(bits per signal change)



NYQUIST LIMIT

Nyquist gives the upper bound for the bit rate of a transmission system

Shows the maximum number of bits that can be sent per second on a *noiseless* channel with a bandwidth of H, if V bits are sent per signal

- Example: what is the maximum data rate for a 13kHz channel that transmits data using 2 levels (binary) ?

▪ **Solution** ($2 \times 13,000 \times \ln 2 = 26,000 \text{ bits/second}$)

EXERCISE

1. Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels.

The maximum bit rate can be calculated as

$$\text{Bitrate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}$$

2. Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits).

The maximum bit rate can be calculated as

$$\text{Bitrate} = 2 \times 3000 \times \log_2 4 = 12000 \text{ bps}$$

INFORMATION THEORY

➤ **Information theory** studies the quantification, storage, and communication of **information**.

➤ Claude E. Shannon in 1948 find **fundamental** limits on signal processing and communication operations such as data compression

➤ Increasing the levels of a signal increases the probability of an error occurring, in other words it reduces the reliability of the system

➤ Shannon's theorem gives the capacity of a channel in the presence of noise

NOISY CHANNEL



We know the past but cannot control it. We control the future but cannot know it.

— Claude Shannon —

Every channel has a capacity C .

—If we transmit at a data rate that is less than capacity, there exists an error control code that provides arbitrarily low BER.

$$C = B \log_2 (1 + S/N)$$

bandwidth of the channel

Channel capacity in bits/s

signal-to-noise ratio

EXERCISE

Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint.

For this channel the capacity C is calculated as

$$C = B \log_2(1 + \text{SNR}) = B \log_2(1 + 0) = B \log_2 1 = B \times 0 = 0$$

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000. The signal-to-noise ratio is usually 3162.

For this channel the capacity is calculated as

$$C = B \log_2(1 + \text{SNR}) = 3000 \log_2(1 + 3162) = 3000 \times 11.62 = 34\,860 \text{ bps}$$

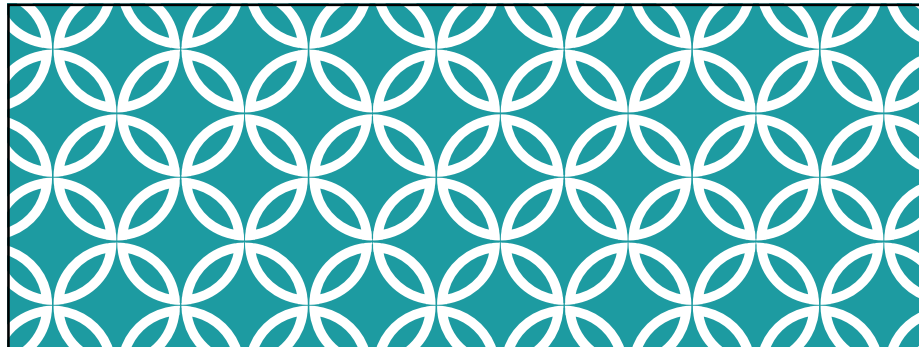
DIFFERENCES B/W NYQUIST & SHANNON

☐ It doesn't take noise into consideration

☐ Shannon's result takes the noise on a line into consideration

☐ Nyquist formula can be used to determine how many signal levels are required to achieve that limit based on how much bandwidth is available

☐ The Shannon limit gives a theoretical maximum limit based on the signal to noise ratio



DIGITAL FILTER

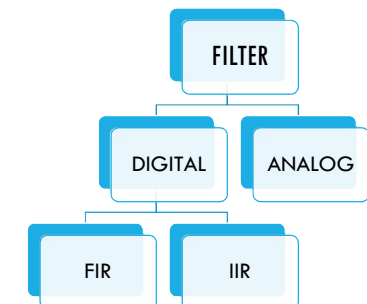
WHY FILTERS?

Two uses :

Signal separation

Signal Restoration

Example: an audio recording made with poor equipment may be filtered to get the original sound.



WHAT IS A FILTER

a **filter** is a device or process that removes some unwanted components or features from a **signal**.

Filtering is a class of **signal processing**

- **Low-pass filter** – low frequencies are passed, high frequencies are attenuated.
- **High-pass filter** – high frequencies are passed, low frequencies are attenuated.
- **Band-pass filter** – only frequencies in a frequency band are passed.
- **Band-stop filter** or band-reject filter – only frequencies in a frequency band are attenuated.
- **Notch filter** – rejects just one specific frequency – an extreme band-stop filter.
- **Comb filter** – has multiple regularly spaced narrow passbands giving the bandform the appearance of a comb.
- **All-pass filter** – all frequencies are passed, but the phase of the output is modified.
- **Cutoff frequency** is the frequency beyond which the filter will not pass signals. It is usually measured at a specific attenuation such as 3 dB.
- **Roll-off** is the rate at which attenuation increases beyond the cut-off frequency.

DIGITAL FILTER SYSTEM

'Digital signal processing consist of anti-aliasing filter ,ADC, digital processor, DAC and a reconstruction filter.

In digital signal processing, there are **two** important types of systems:

- **Digital filters:** perform signal filtering in the time domain
- **Spectrum analyzers:** provide signal representation in the frequency domain

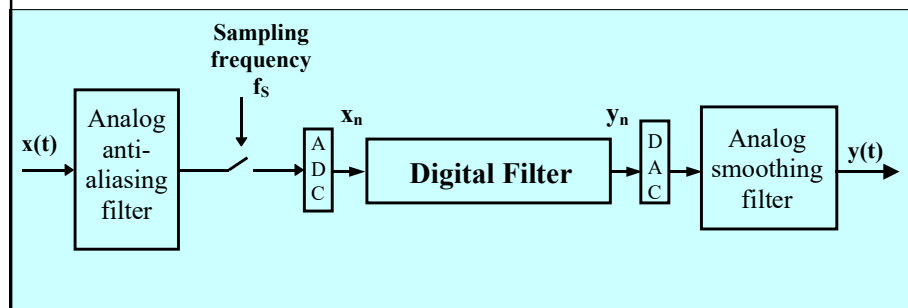
A DIGITAL FILTER

A **digital filter** is a system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal.

Digital filters are used for two general purposes:

- (1) **separation** of **signals** that have been combined,
- (2) **restoration** of **signals** that have been distorted in some way

DIGITAL FILTER:



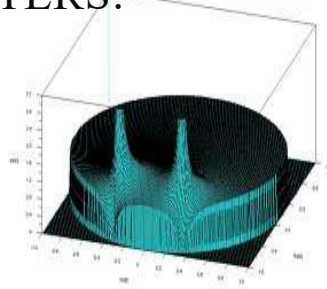
ADVANTAGES OF FILTERS:

DSP filter is immune to:

- Environmental Changes
- Noise and relatively Stable
- Impedance Matching
- Computational Problems

Availability of :

- Multiple Filtering
- Variety of Shapes for Amplitudes and Phase Response.
- Easy Transportation and Reconfiguration.



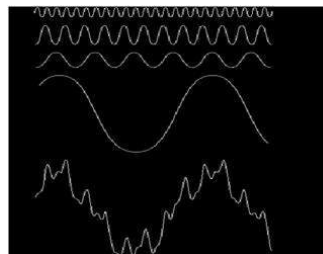
FOURIER TRANSFORM

FOURIER SERIES

Fourier series: periodic signals can be represented into sum of sines and cosines when multiplied with a certain weight.

It further states that periodic signals can be broken down into further signals with the following properties.

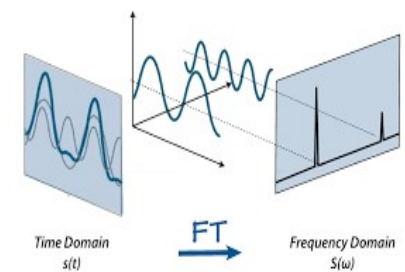
- The signals are sines and cosines
- The signals are harmonics of each other



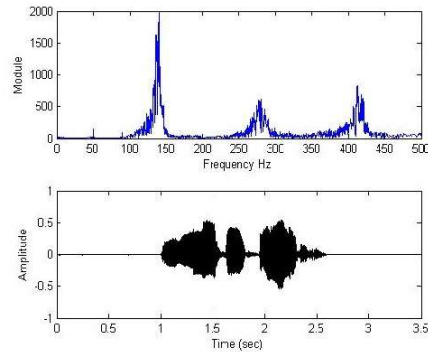
THE FOURIER TRANSFORM

The Fourier transform: the non periodic signals whose area under the curve is finite can also be represented into integrals of the sines and cosines after being multiplied by a certain weight.

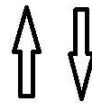
- The **Fourier transform** decomposes a function of time (a signal) into the frequencies
- The **Fourier transform** is called the frequency domain representation of the original signal.



WHY WE NEED TRANSFORM



Frequency Domain



Fourier Transform

Time Domain

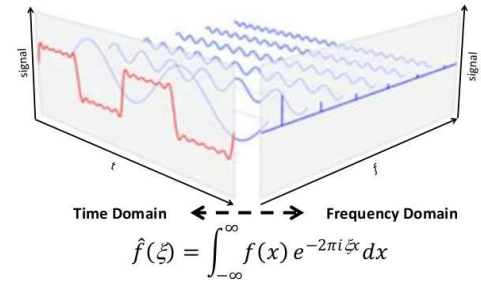
Fourier was a mathematician in 1822. He gave Fourier series and Fourier transform to convert a signal into frequency domain.

Jean-Baptiste Joseph Fourier



$$x(\omega) = \int_{-\infty}^{\infty} x(t) \cdot e^{-j\omega t} dt$$

Fourier Transform - Review

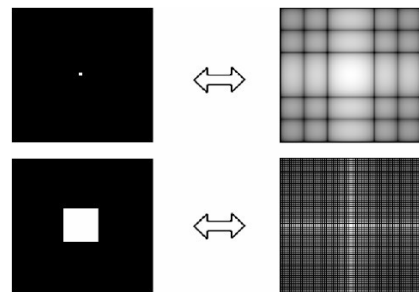


Time Domain \longleftrightarrow Frequency Domain

$$\hat{f}(\xi) = \int_{-\infty}^{\infty} f(x) e^{-2\pi i \xi x} dx$$

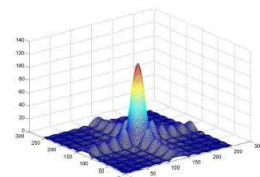
EXAMPLE OF FT

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$



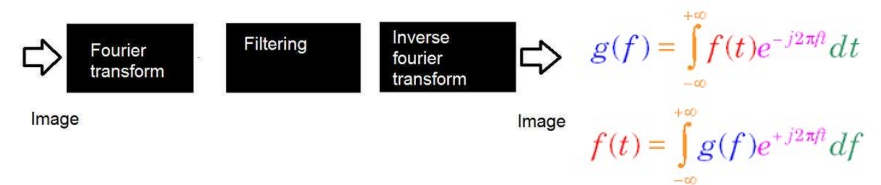
Input images

FFT



3D view of FFT

INVERSE FOURIER TRANSFORM



$$g(f) = \int_{-\infty}^{+\infty} f(t) e^{-j2\pi ft} dt$$

$$f(t) = \int_{-\infty}^{+\infty} g(f) e^{j2\pi ft} df$$

DIFFERENCE BETWEEN FOURIER SERIES AND TRANSFORM

Although both Fourier series and Fourier transform are given by Fourier, but the difference between them is Fourier series is applied on periodic signals and

Fourier transform is applied for non periodic signals

USING DIFFERENT METHODS

IMPORTANT TOPIC

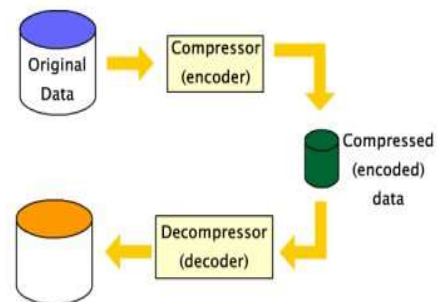
CONVOLUTION

- LAPLACE TRANSFORM
- CONVOLUTION FORMULA
- Z TRANSFORM
- FOURIER TRANSFORM

*• signal and system
• dtsp
• dsp*

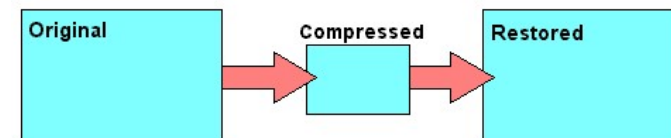
DATA COMPRESSION

Data compression is particularly useful in communications because it enables devices to transmit or store the same **amount of data** in **fewer bits**

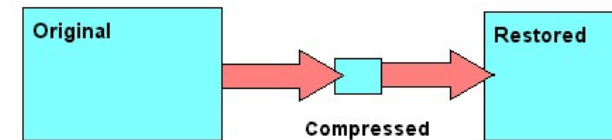


DATA COMPRESION

LOSSLESS



LOSSY



ADVANTAGES OF DSP:

- ❑ Accuracy
- ❑ Flexibility
- ❑ Easy operation
- ❑ Multiplexing
- ❑ Storable

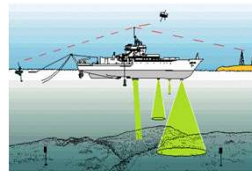
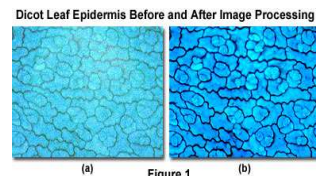
LIMITATIONS OF DSP

- ❑ Antialias filter
- ❑ Frequency Resolution
- ❑ Quantization Error



APPLICATIONS OF DSP:

- ❖ Image processing
- ❖ Consumer application
- ❖ Cellular mobile phones
- ❖ In Communication
- ❖ In Speech and Music
- ❖ In Biomedical
- ❖ In Radar and Sonar



CONCLUSION :

- Digital Processing – a **series of instructions** to manipulate the **digital numbers**.
- DSP is used in wide range in everyday applications.
- Digital Signal Processing (DSP) is often used in modern audio logical equipment.
- Storage capability.

