# Introduction to DSP

by

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# DSP AROUND YOU


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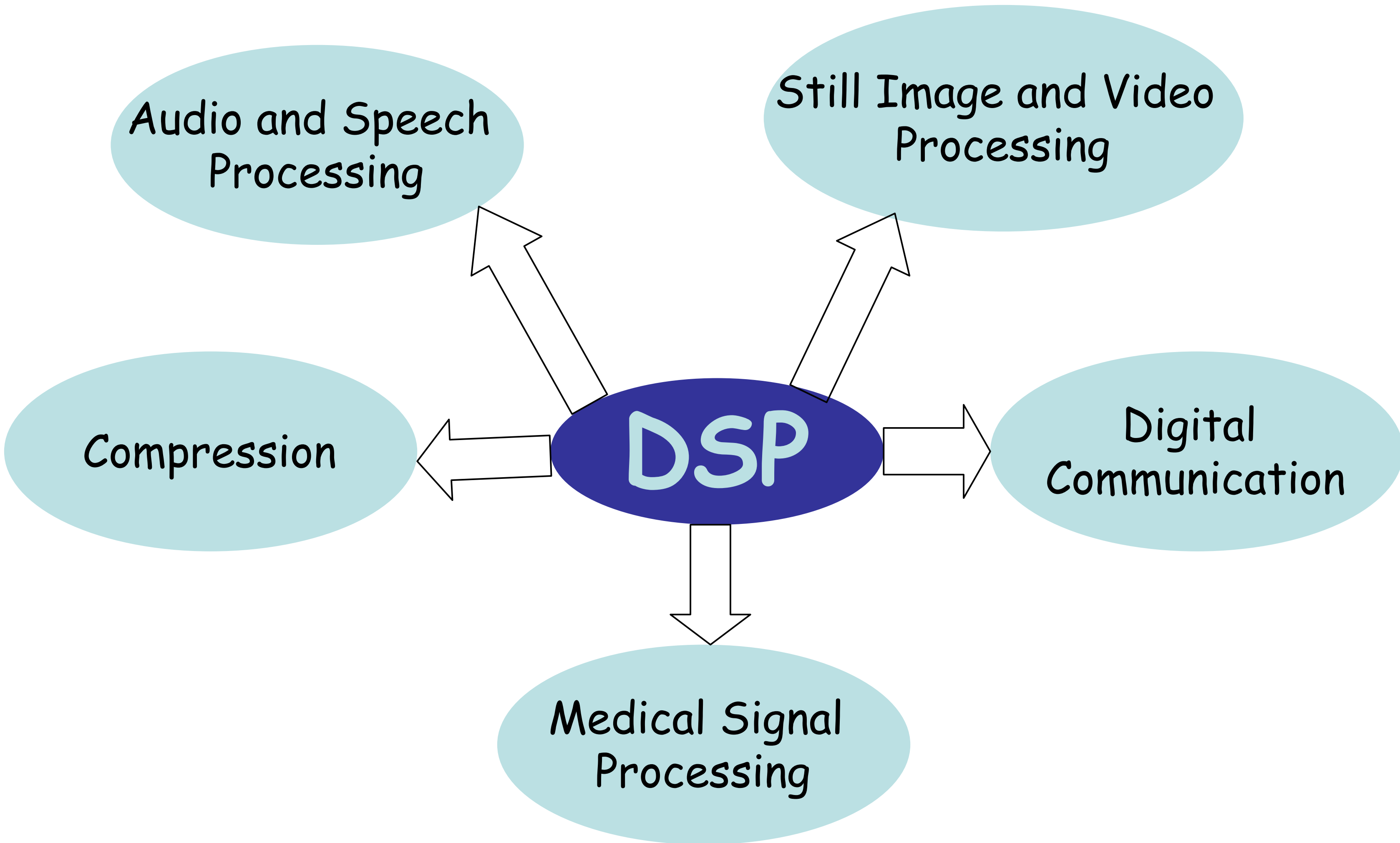
# Core Areas

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## What is DSP ?

DSP is the **mathematics**, the **algorithms**, used to manipulate the signals after they have been converted into a digital form.

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## Goals of DSP ?

- **enhancement** of signals
- **extraction** of **information**
- **compression** of data for storage and transmission

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# Applications of DSP

## Telecommunications

- Signaling tone generation and detection
- Multiplexing
- Echo control
- Compression

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# Speech and Audio Signal Processing

- Music
- Speech Generation and Recognition
- Compression
- Echo control

How many bits are required to be transmitted per second when a speech signal is sampled at 8 Khz, and 16-bits are used to represent one sample?

**128000 bits**

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# Image Processing

- Image processing
- Image Compression
- Video Processing
- Video Compression

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# Some more Applications

- Radar
- Sonar
- Seismology
- Medical
- Industrial

# Why Digital ???

- Once converted to numbers the signal is unconditionally stable

- **Simplicity**

– some things can be done more easily digitally than with analog systems

- Flexibility

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# Why Digital??? (Contd.)

- **Versatility**

- digital systems can be reprogrammed for other applications
- digital system can be ported to different hardware (for example a different DSP chip or board level product)

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# Why Digital??? (Contd.)

- **Repeatability**

  - digital system can be easily duplicated
  - digital systems do not depend on strict component tolerances

- **Easy storage**

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## Why Digital??? (Contd.)

- Off-Line Processing
- Security can be improved by encrypting the numbers

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Signal: An entity that bears information.

Or

A signal is defined as a physical quantity that varies with time, space or other independent variable/s.

System: A system is defined as a physical device or an algorithm that performs an operation on the signal.

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Signal processing: When certain operations are performed on the signal and the latter undergoes some change, it is called signal processing.

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# Analog signal Processing System


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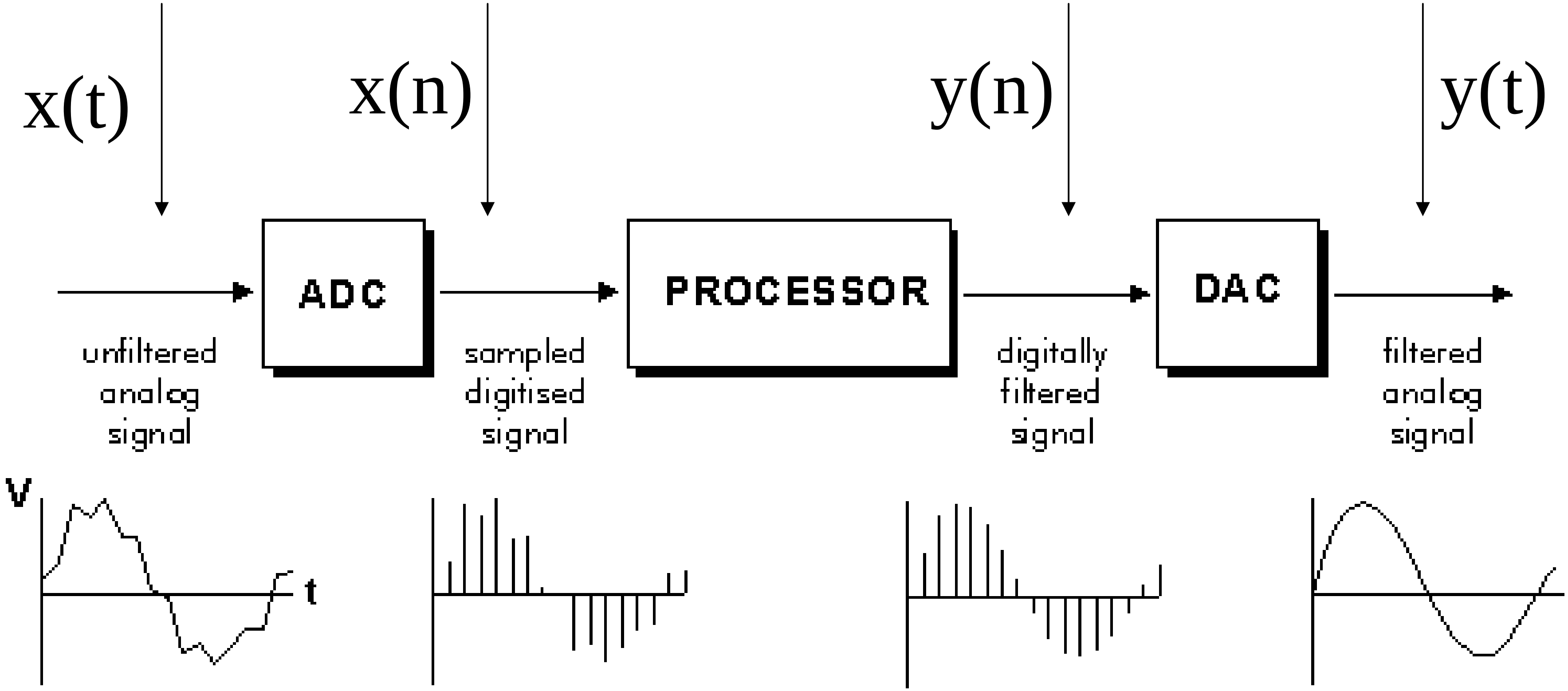
# Block Diagram of a DSP System

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# Steps

- Acquire the Signal
- Convert it to a sequence of numbers
- ***Process the numbers as required***
- Transmit or save the data (numbers) as required
- Convert the processed sequence of numbers back to a signal

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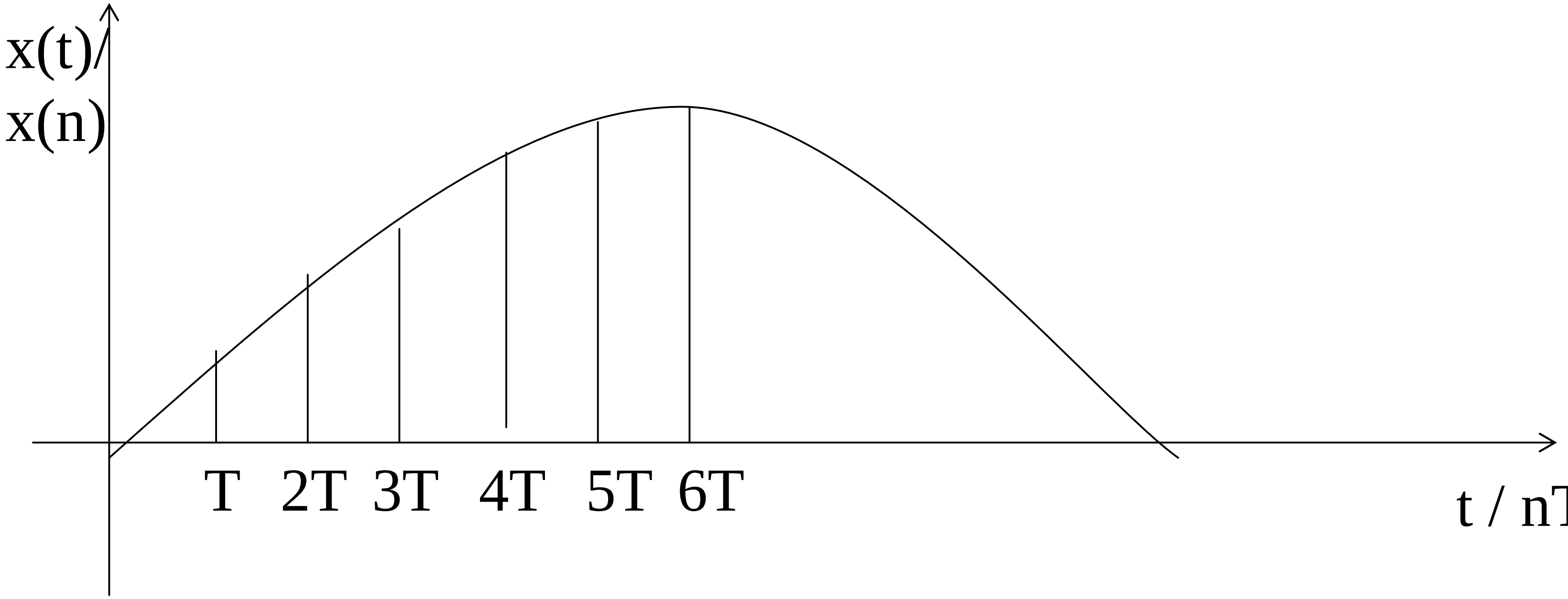
# Sampling




$$x(n) = x(nT) \qquad t = nT$$

$$F_s = 1/T \qquad t = n/Fs$$

$$x(n) = A^* \text{Cos}(2\Pi fn) \quad \text{or} \quad x(n) = A^* \text{Cos}(\omega n)$$

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# Problem : Aliasing

If $F < F_s/2$, $f < 1/2$ ----- Fundamental frequency range

Frequencies outside fundamental range, after sampling, are indistinguishable from frequency F, and hence they are **aliases** of F.

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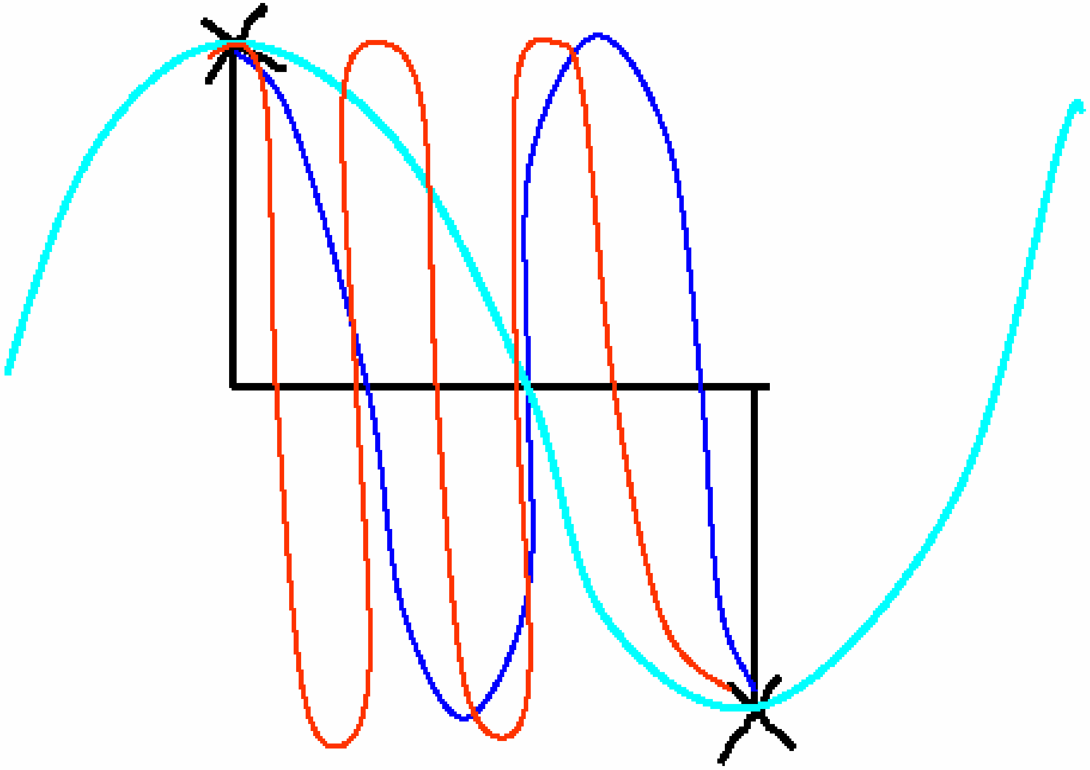
# Problem



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# Aliasing

- Alias means assuming some other identity.

- Phenomenon of C.T. signal of **higher frequency** acquiring identity of **lower frequency** after sampling is called **aliasing.**

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# Sampling Theorem

If the highest frequency component in a signal is $F_{max}$, then the signal should be sampled at the rate atleast 2 $F_{max}$ for the samples to describe the signal completely.

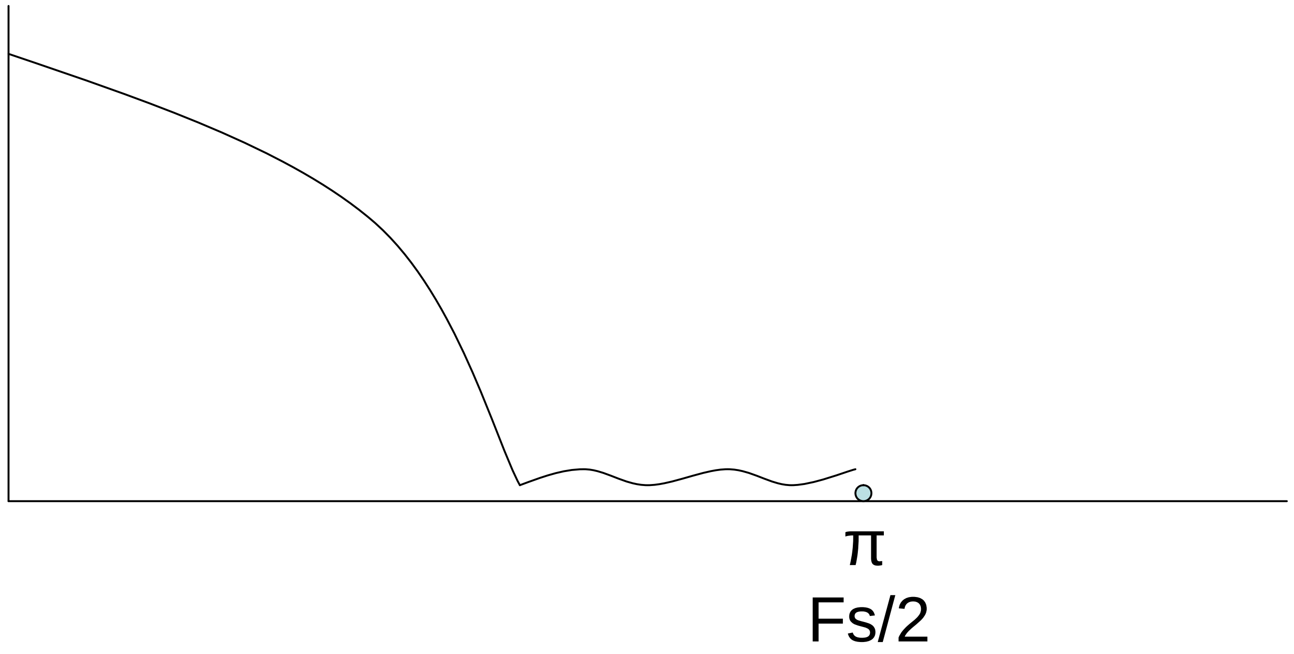
$$F_s >= 2\ F_{max}$$

where $F_s$ is the sampling frequency.

$F_s/2$ is called **folding frequency** or **Nyquist rate**

Range of digital frequencies is from 0 to pi

Analog equivalent is from 0 to $F_s/2$

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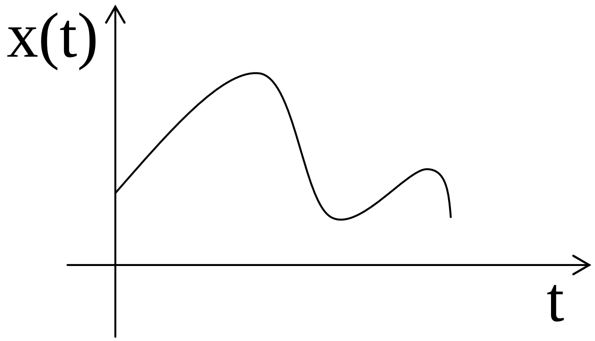
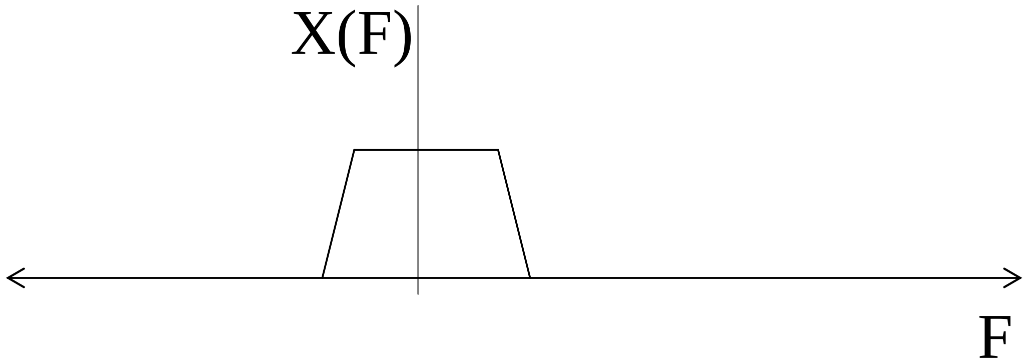
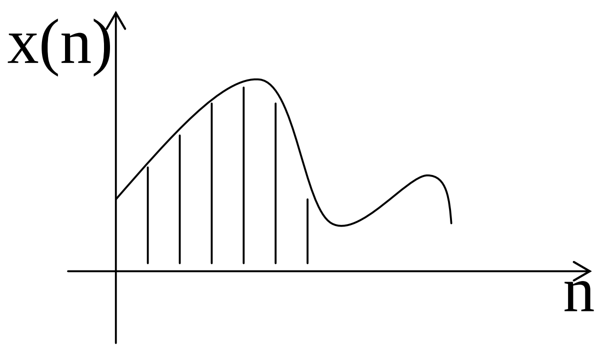
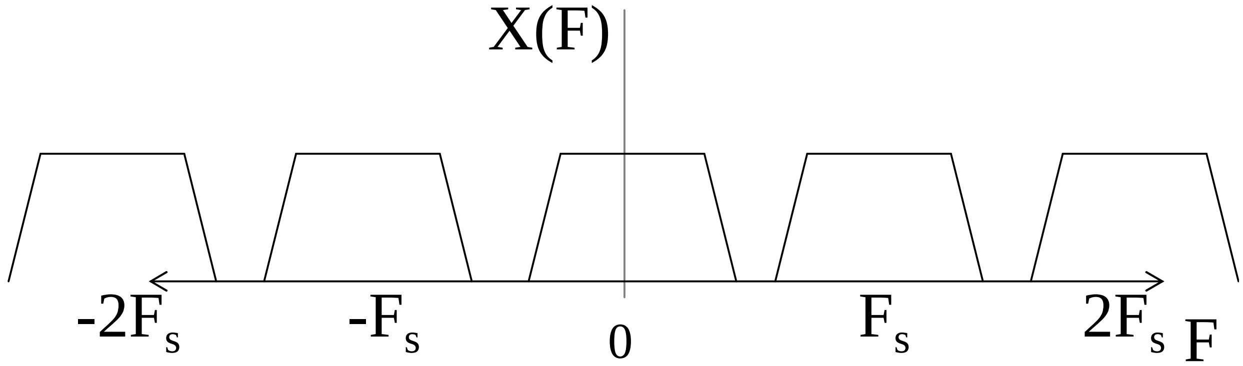
# Signal and its Spectrum

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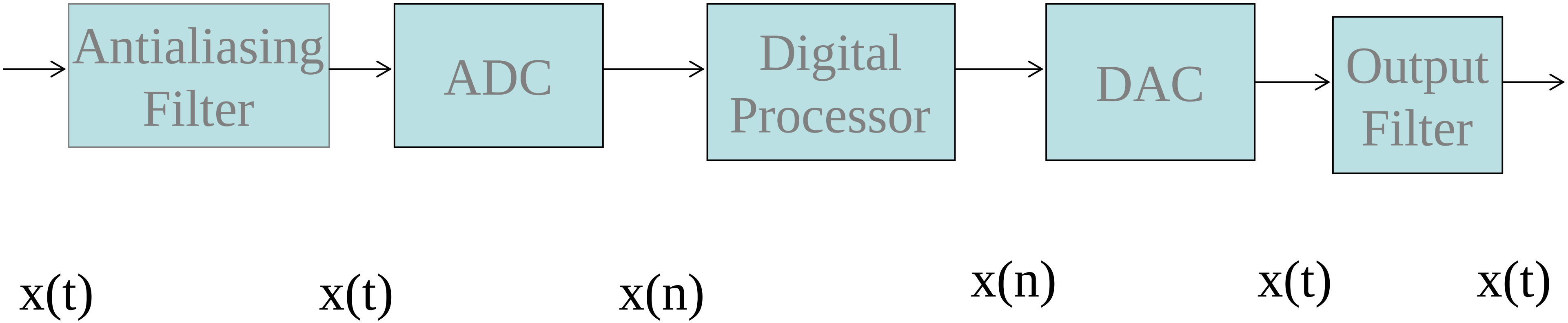
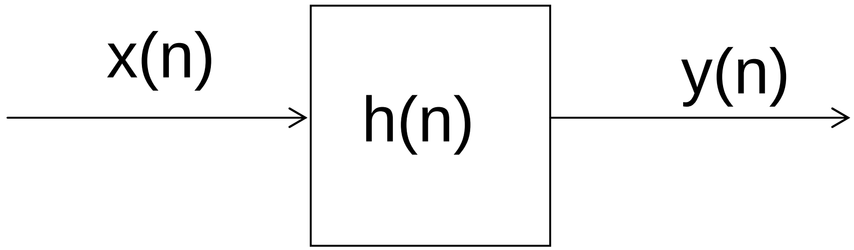
# Real Time DSP System




Antialiasing filter is a sharp cut-off analog low pass filter used to bandlimit the signal.

It should provide sufficient attenuation at high frequencies.

# Convolution




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

where h(n) is the impulse response of the filter

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# Real Time Convolution

$h(n)=\{h_0\ h_1\}$  $x(n)=\{x_0\ x_1\ x_2\ x_3\ x_4\ x_5\ x_6\ x_7\ x_8\ x_9\ x_{10}\ x_{11}\ldots\ldots\}$

$y0 = x_{-1}h_1 + x_0h_0$

$y(1)= x_0h_1 + x_1h_0$

$y(2)= \underline{x_0h_2} + x_1h_1 + x_2h_0$

$y(3)= x_2h_1 + x_3h_0$

$y(4)= x_3h_1 + x_4h_0$

| x | x | x | x | x | x | h |
|---|---|---|---|---|---|---|
| 0 | $x_0$ | $x_1$ | $x_2$ | $x_3$ | $x_4$ | $h_0$ |
| 0 | 0 | $x_0$ | $x_1$ | $x_2$ | $x_3$ | $h_1$ |

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# Issues with h[n]

- **Stability**
- Bounded Input Bounded Output
- Hence, h[n] should be absolutely summable.

- **Causality**
- Implementation should be possible
- $h[n] = 0$; for $n < 0$.

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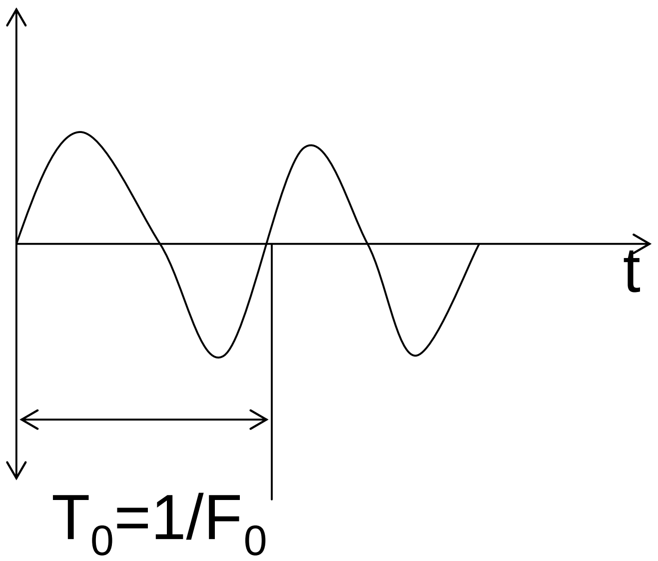
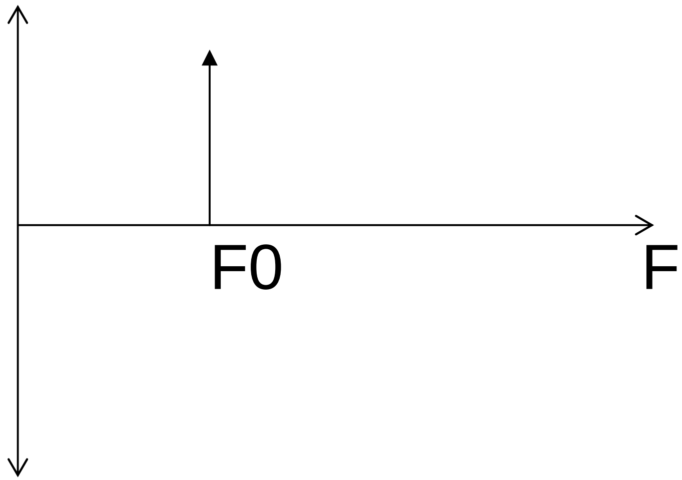
# Time and Frequency

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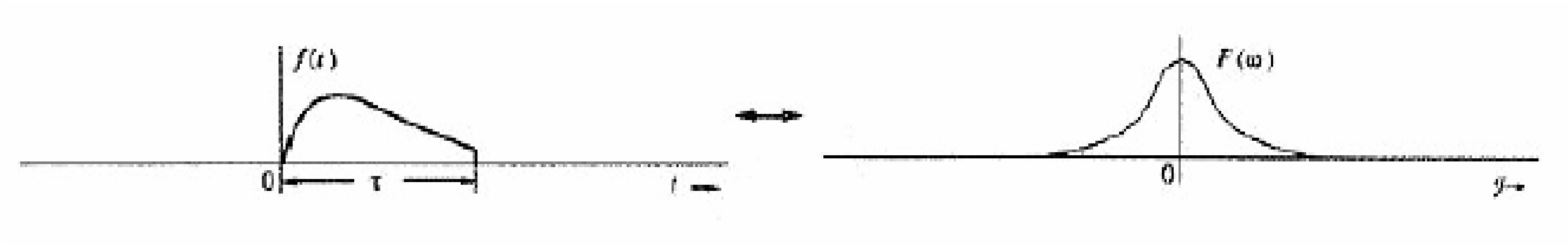
# Fourier Transform

## Continuous Time Case :

$$H(F) = \int_{-\infty}^{+\infty} h(t) e^{-j2\Pi Ft} dt$$

$$h(t) = \int_{-\infty}^{+\infty} H(F) e^{j2\Pi Ft} dF$$

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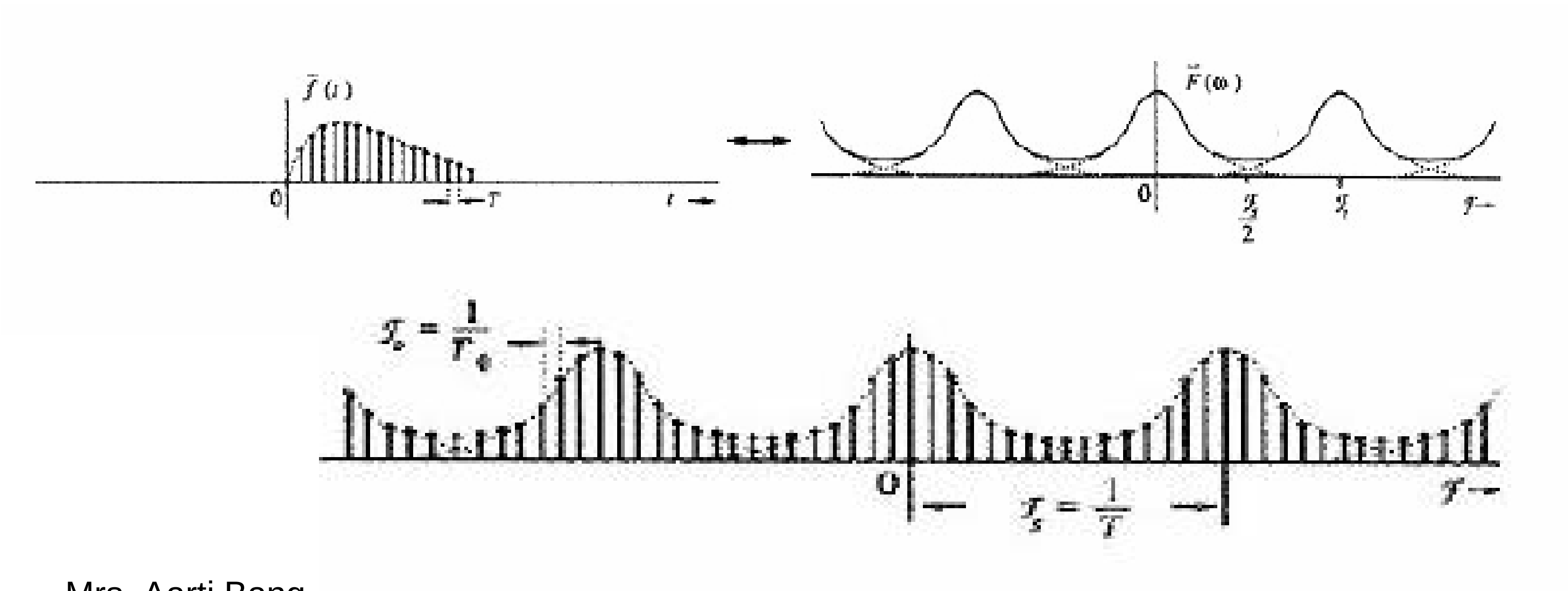
# Discrete Time Case :

$$H(\omega) = \sum_{n=-\infty}^{+\infty} h(n) e^{-j\omega n}$$

$$h(n) = \frac{1}{2\pi} \int_{-\pi}^{+\pi} H(\omega) e^{j\omega n} d\omega$$



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# Discrete Fourier Transform

**DFT**

$$H(k) = \sum_{n=0}^{N-1} h(n) e^{-j2\pi nk/N} \qquad k = 0,1,2.....N-1$$

**IDFT**

$$h(n) = \frac{1}{N} \sum_{k=0}^{N-1} H(k) e^{j2\pi nk/N} \qquad n = 0,1,2.....N-1$$

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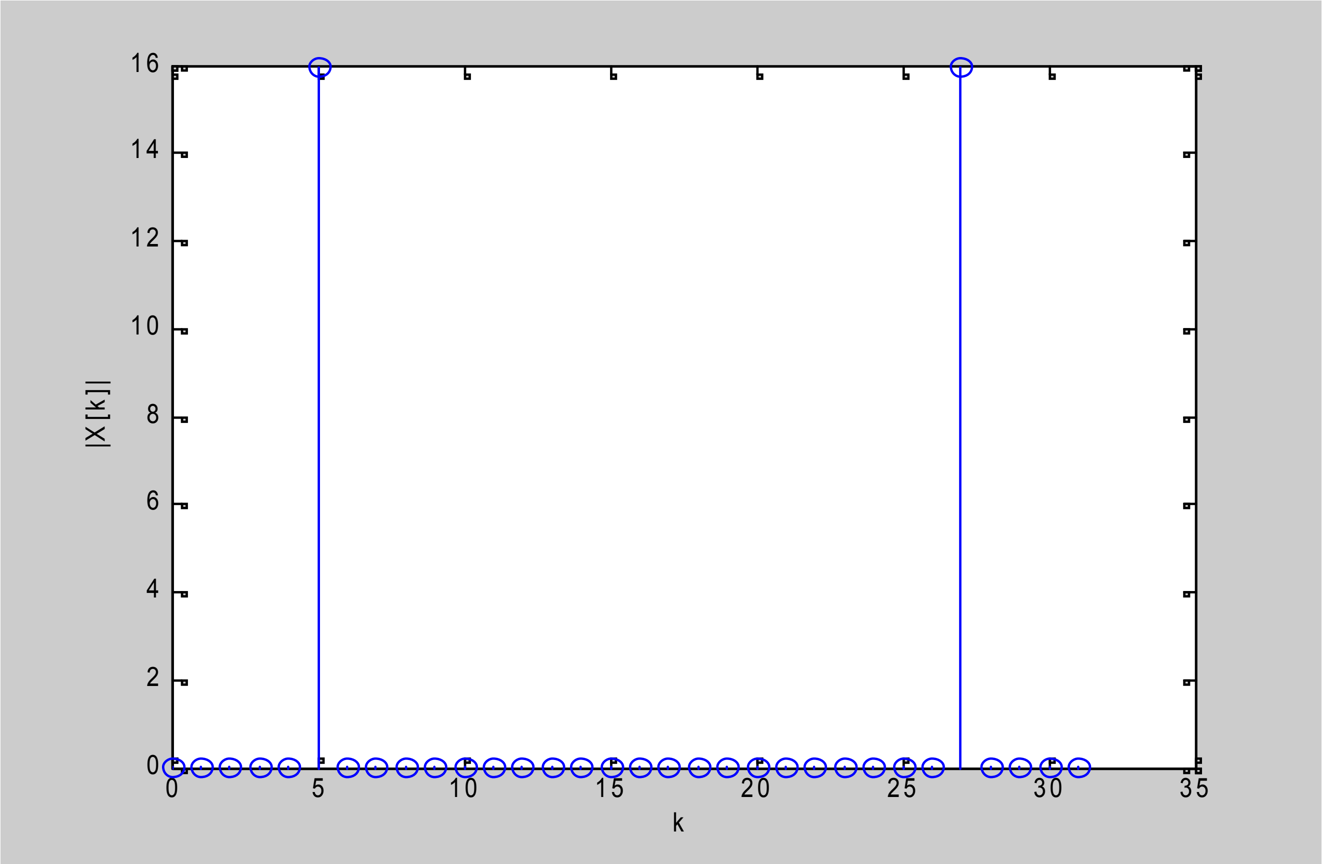
**e.g.**

F=10 Hz    Fs = 64 Hz    N = 32

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$$H(\omega) = \sum_{n=-\infty}^{+\infty} h(n) e^{-j\omega n}$$

$$\omega = \frac{2\pi k}{N}$$

$$H(k) = \sum_{n=0}^{N-1} h(n) e^{-j2\pi nk/N}$$

$$\frac{2\pi F_k}{Fs} = \frac{2\pi k}{N}$$

$$F_k = \frac{kF_s}{N}$$

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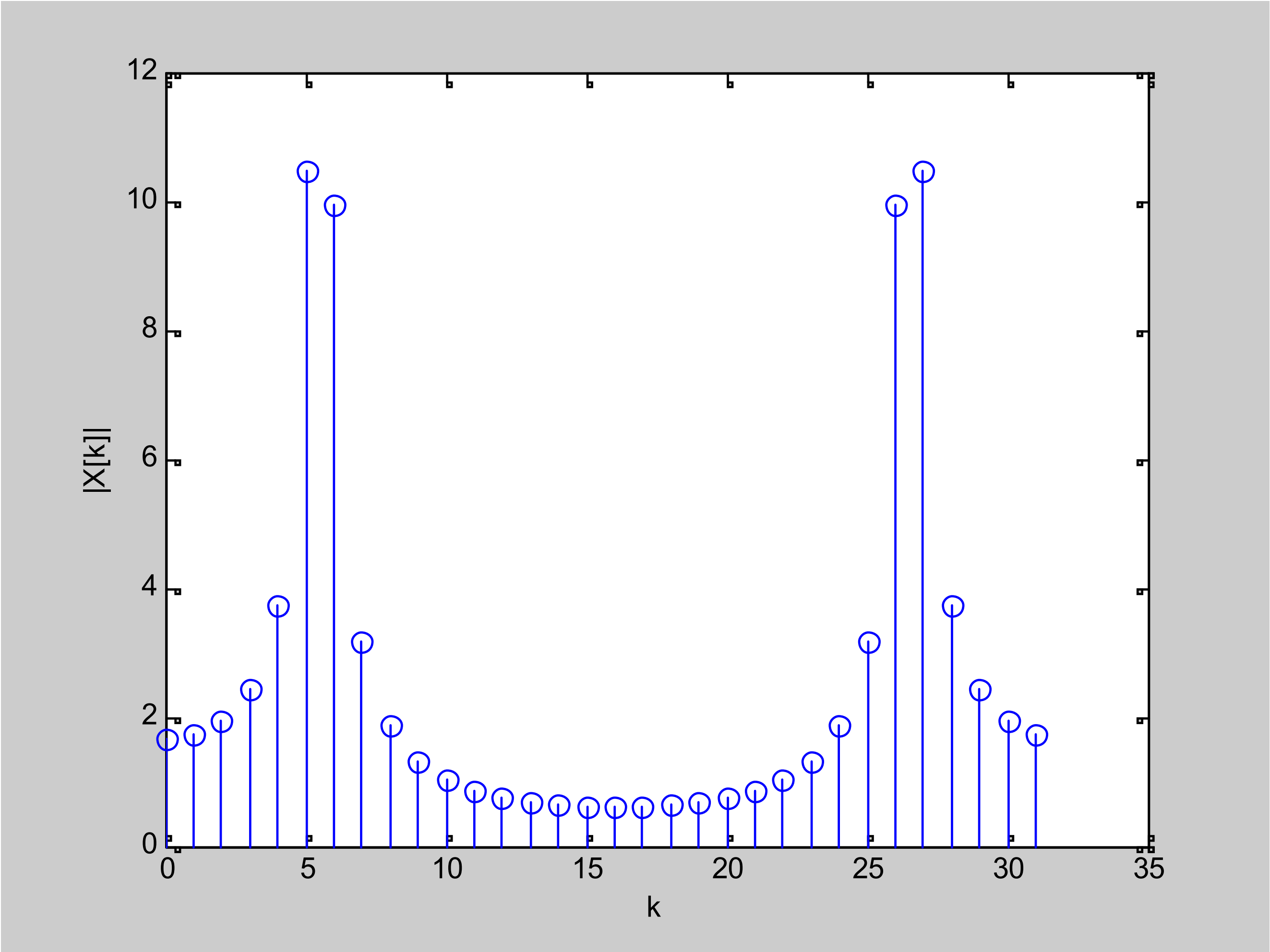
# F=11 Hz     Fs = 64 Hz     N = 32




This phenomenon is called **leakage**

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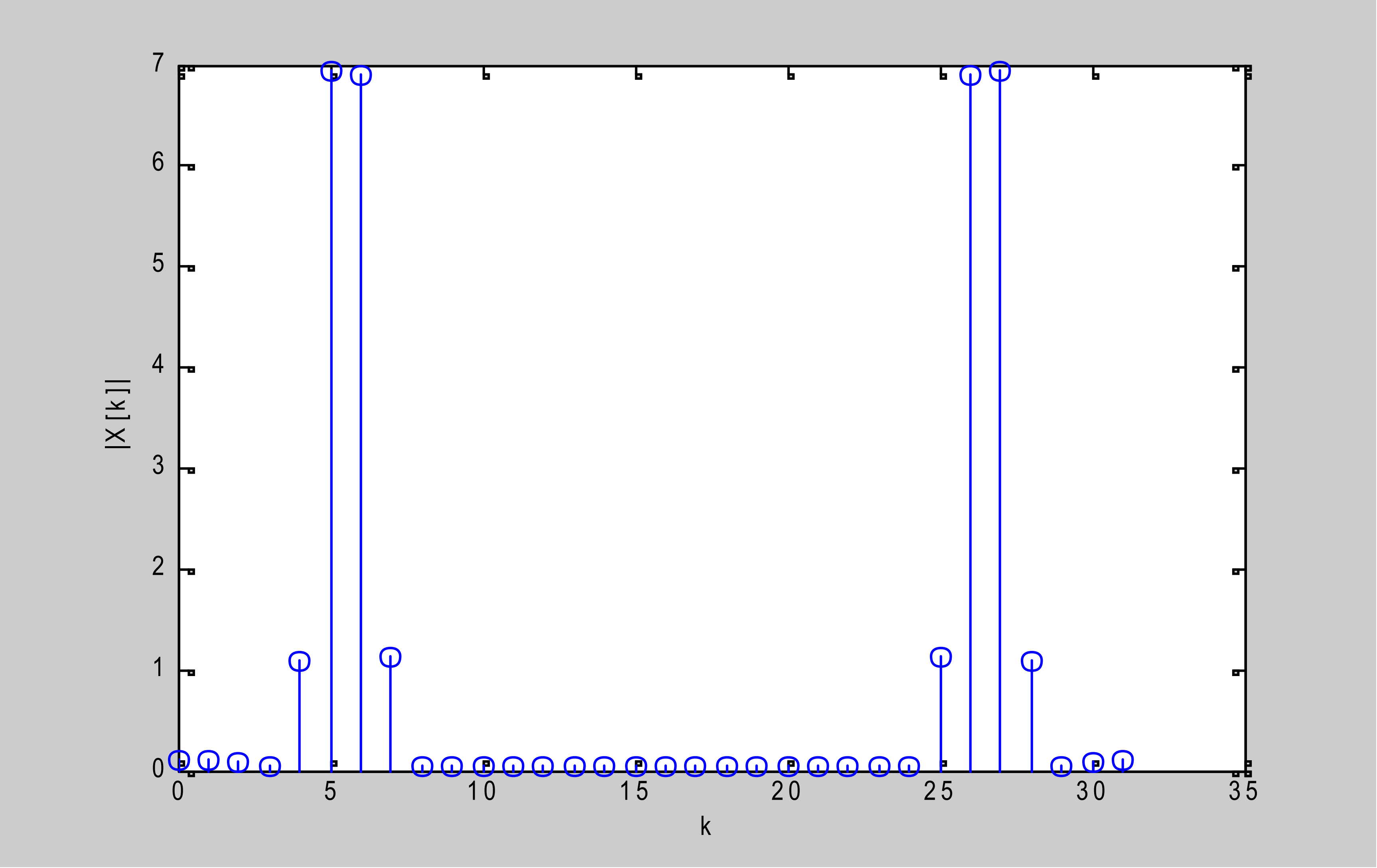


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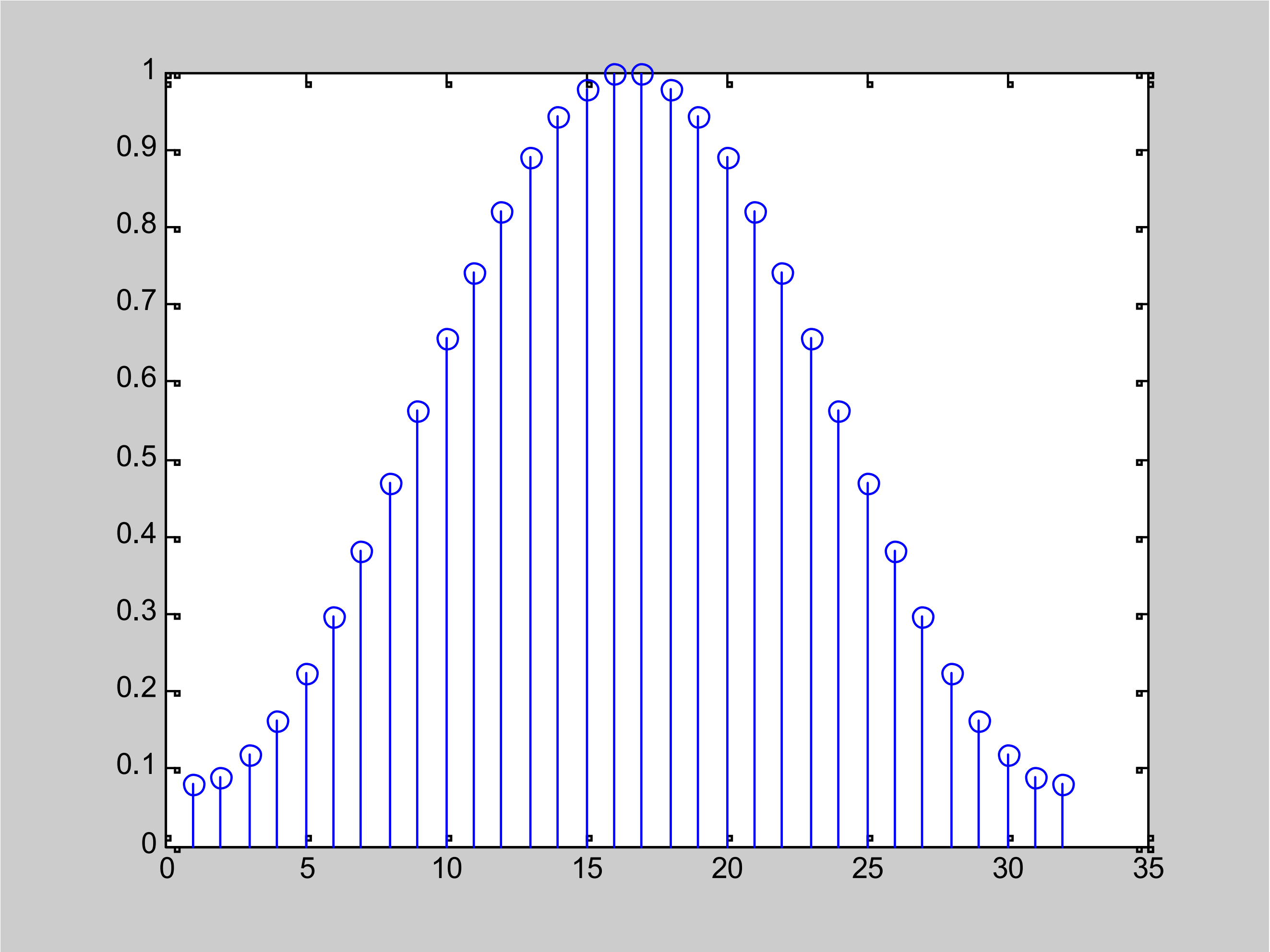
# Hamming Window Function

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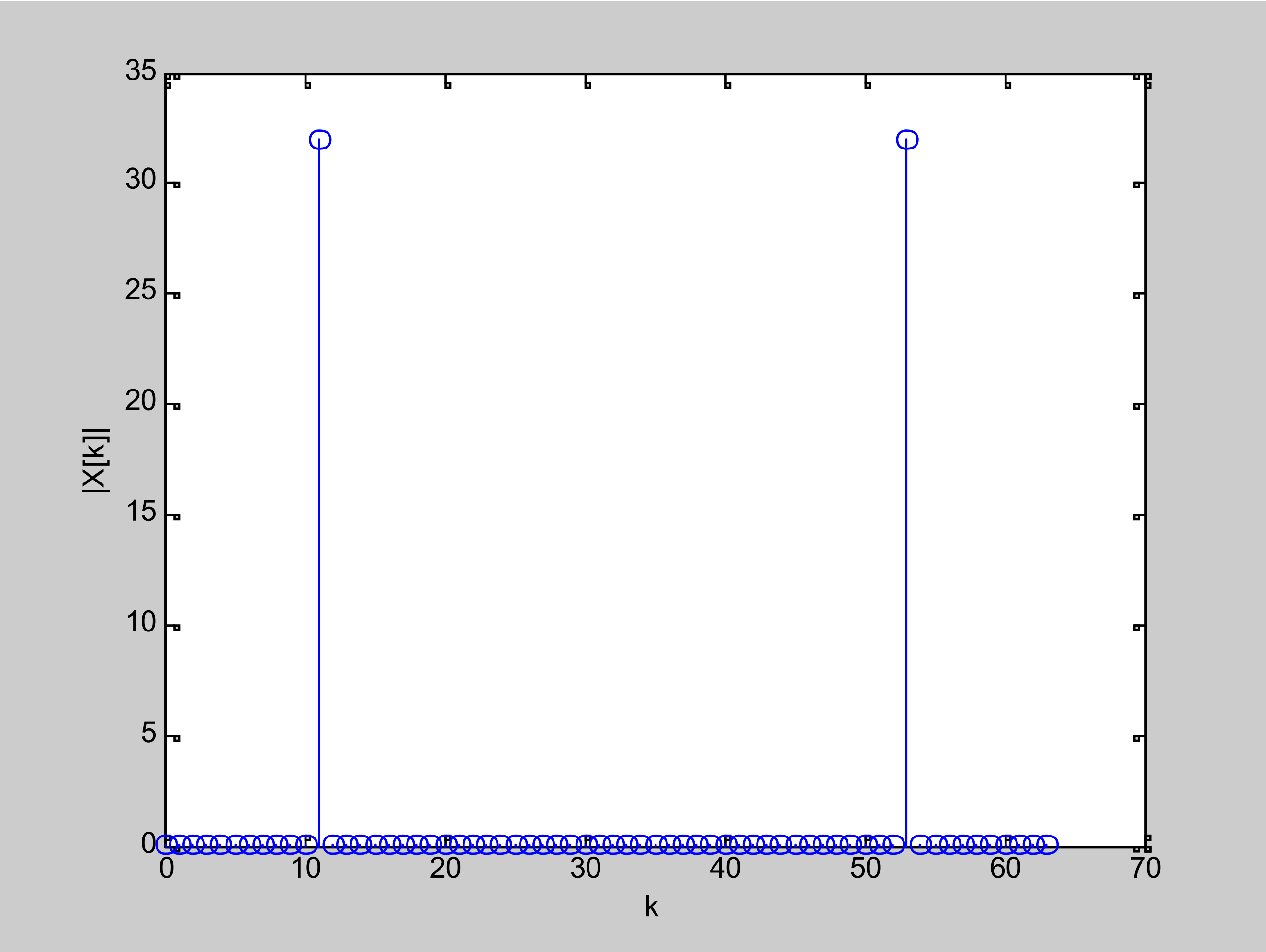
# F=11 Hz Fs = 64 Hz N = 64

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# Digital Filters

- Infinite Impulse Response (IIR) Filter

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

- Finite Impulse Response (FIR) Filter

$$y(n) = \sum_{k=0}^{M} b_k x(n-k)$$

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# Any Questions ?