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Lecture 2- Analog to Digital Convertor Asst. Prof. Dr. Mehdi Ebady Manaa



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# Analog to digital converter



The connection of digital circuit to sensing device can be done only if the sensors are inherently digital themselves. However, when analog signals are involved in the project, the interface becomes much more complex. In this case, it needs a way to translate analog signals into digital form: an ADC; digital-to-analog converter or DAC performs the opposite operation.



# What is ADC?

• ADC (Analog to Digital Converter) is an electronic device that

converts a continuous analog input signal to discrete digital numbers (binary)

o Analog

Real world signals that contain noise

- : Continuous in time
- o Digital

Discrete in time and value Binary digits that contain values 0 or 1



# Why is ADC Important?

- All microcontrollers store information using digital logic
- o Compress information to digital form for efficient storage
- Medium for storing digital data is more robust
- Digital data transfer is more efficient
- o Digital data is easily reproducible
- Provides a link between real-world signals and data storage

## How ADC Works

- 2 Stages:
- o Sampling
  - Sample-Hold Circuit
  - o Aliasing
- o Quantizing and Encoding
  - Resolution





ADC block diagram.

- •
  - Reduction of a continuous signal to a discrete signal
  - <sup>o</sup> Achieved through sampling and holding circuit
  - Switch ON sampling of signal (time to charge capacitor w/ Vir)
  - Switch OFF- voltage stored in capacitor (hold operation)
  - <sup>0</sup> Must hold sampled value constant for digital conversion





Conversion of a continuous-time signal to discrete time.



Photographs in newsprint are "half-tone- images. Each point is black or white and the average conveysbrightness.





Zoom in to see the binary pattern.



### The sampling theorem

The sampling theorem, or more correctly Shannon's sampling theorem, states that we need to sample a signal at a rate at least twice the maximum frequency component in order to retain all frequency components in the signal. This is expressed as

#### $f_{\rm s} > 2f_{\rm max},$

where **/S** is the sampling rate (frequency), !max is the highest frequency in the input signal, and the minimum required rate (2/maJ is called the Nyquist frequency.

The time interval between the digital samples is

$$\Delta t = \frac{1}{f_{\rm s}}.$$