Data Acquisition

2103-602

Measurement and

Instrumentations

Outline I

- Digital Data Acquisition and its function
- Data Sampling
- Quantization: A/D, D/A converters

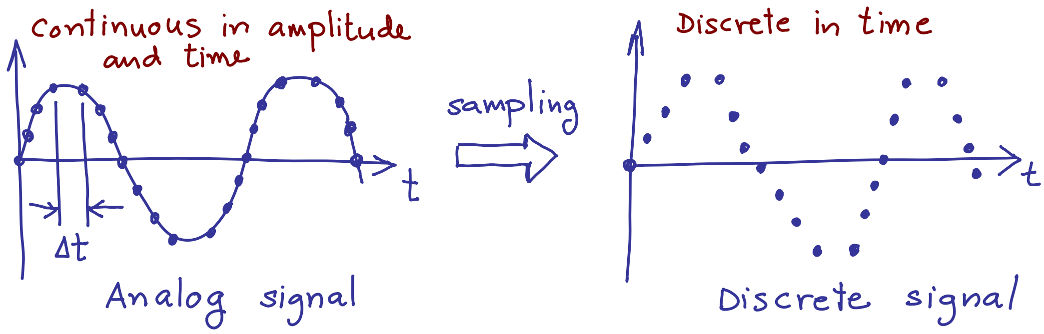
How data acquisition functions

The hardware receives an analog signal and converts it to digital signal, so called digitize, ready for processing by the computer.

Digitization of the analog signal requires two separate operations:

– **Sampling**: define the points or rate at which the data is acquired. Then loss time continuity

Data Sampling



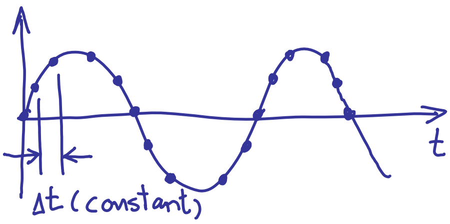
How accurately the discrete signal represents the analog signal depends on:

– Frequency content of analog signal

– Time increment Δt between each discrete number, or a sampling rate

– Total sampling time of measurement

Sampling Rate



If Δt is the time increment or time resolution, the sampling rate or sampling frequency (in Hz) is defined as

$$\boxed{\Delta t \downarrow \;\Rightarrow\; f_s \uparrow}$$

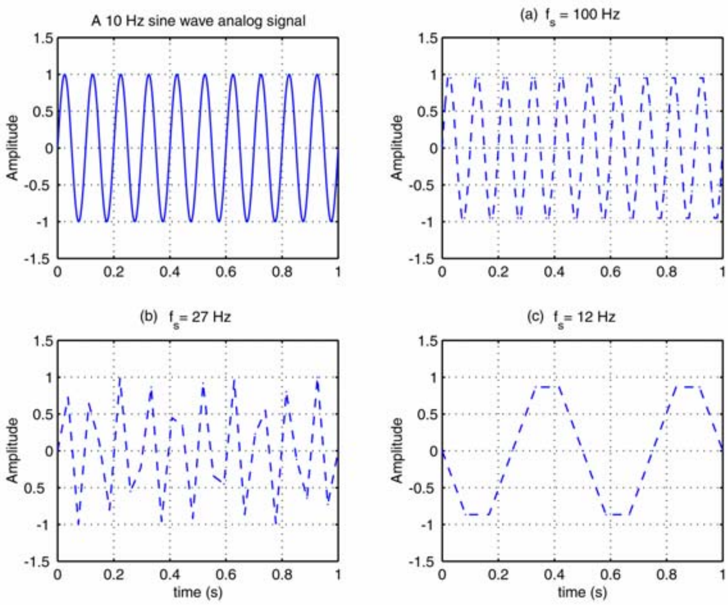
$$f_s = \frac{1}{\Delta t}$$

Sampling Theorem

Sampling rate must be more than twice of the highest frequency contained in the measured signal.

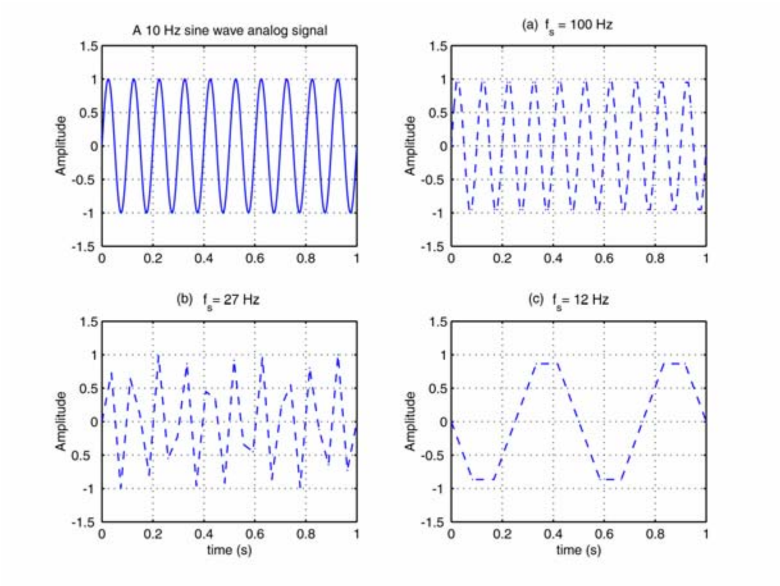
$$\boxed{\begin{array}{l} f_s > 2 f_m \\ \text{or } \; \Delta t < \dfrac{1}{2 f_m} \end{array}}$$

(f_m ~ maximum frequency in analog signal)



Aliasing

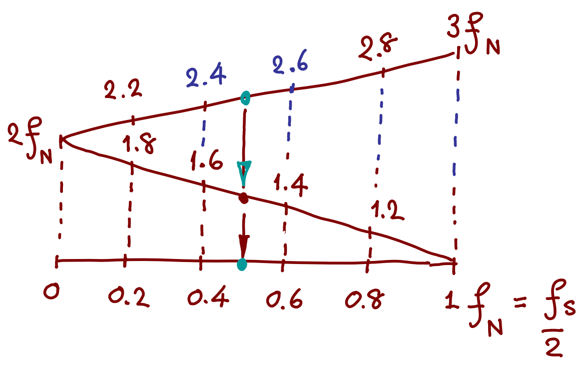
A misinterpretation of the frequency contents, occurring when the sampling rate does not satisfy the sampling theorem, e.g. case c



Nyquist frequency

Define the Nyquist frequency as $f_n = f_s/2 = 1/(2\,\Delta t)$

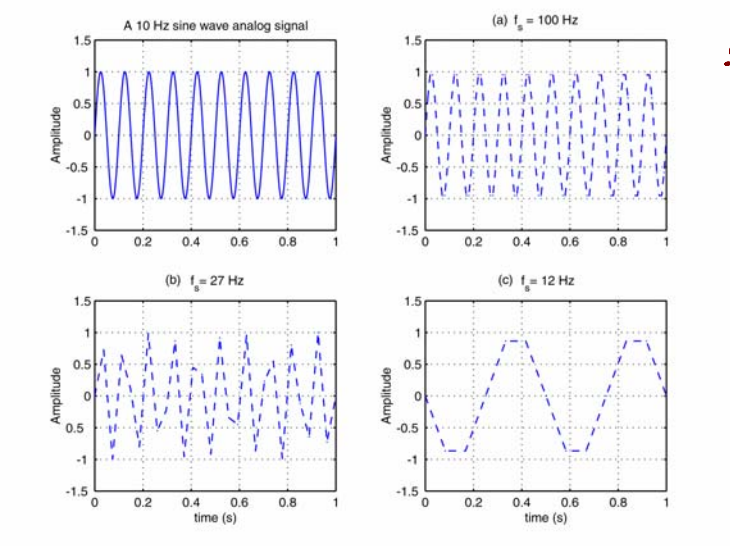
f_n represents a folding point for aliasing phenomena. All frequency contents in analog signal whose frequency are higher than f_n will appear at the alias frequencies less than f_n.

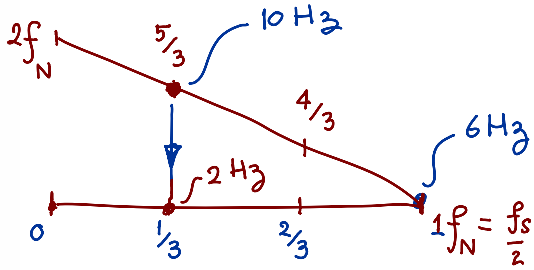


E.g. $f = 0.5f_N$, $1.5f_N$, $2.5f_N$ will appear as aliasing frequency of $0.5f_N$

(No problem if: $f_s > 2f_m$)

Example: case (c)



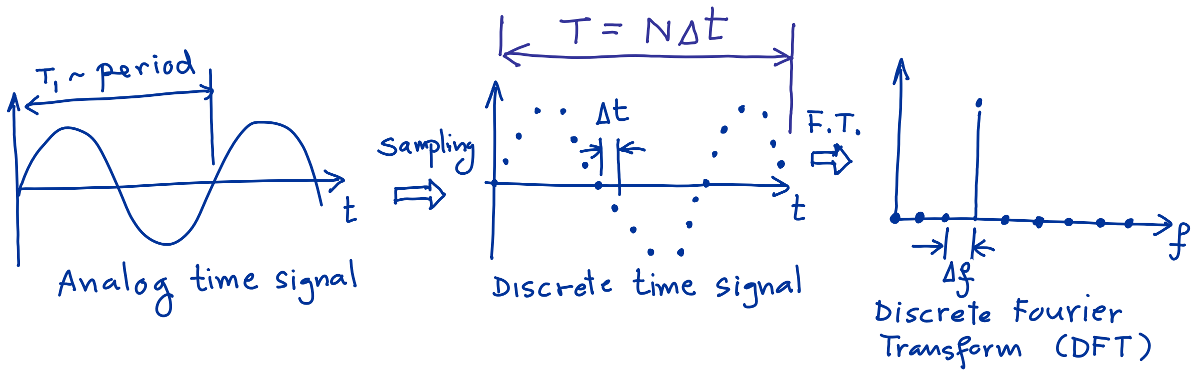


If f_s = 12 Hz then f_n = 6 Hz which is less than the original frequency of 10 Hz. Thus we see an alias frequency of 2 Hz from this sampling.

Q: How to avoid the alias phenomena when sampling a signal of unknown frequency content?

A: Pre-filtering the signal with low-pass or anti-aliasing filter before the sampling.

Discrete Fourier transform (DFT)



In practice, we determine Fourier transform of an analog signal by using fast Fourier transform (FFT). The DFT then represents its Fourier transform.

Q: Is DFT accurately represented the Fourier transform?

Leakage (Amplitude ambiguity)

The DFT is accurate iff:

① Total sampling time

$T = N\Delta t = mT_1$

where $m \sim$ integer

② $\Delta t < \frac{1}{2f_m}$ or $f_s > 2f_m$

(sampling theory)

where:

$\Delta t \sim$ time increment

$N \sim$ # of sample points

$\Delta f \sim$ frequency resolution

$T_1 \sim$ fundamental frequency

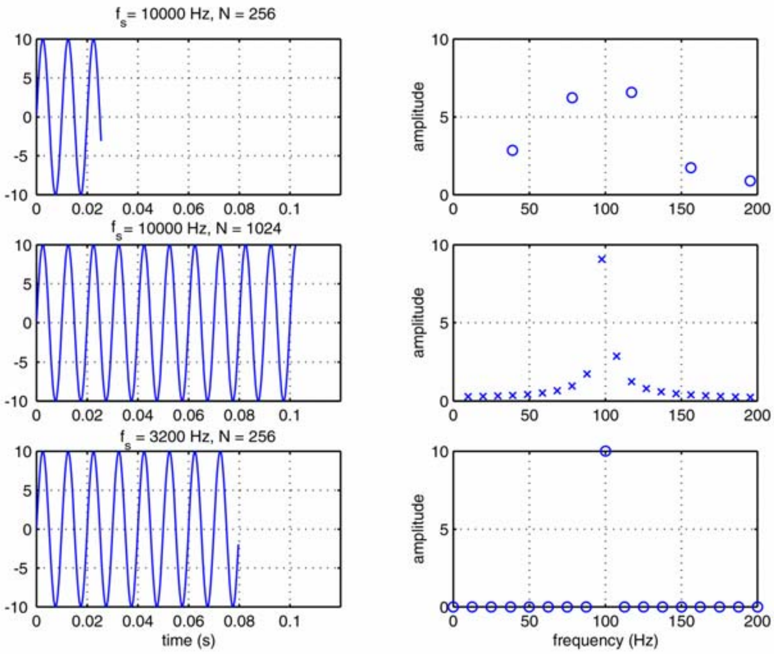
$$\Delta f = \frac{1}{N\Delta t} = \frac{f_s}{N}$$

If condition 1) is unsatisfied, DFT will have a *leakage* of *amplitude* to adjacent frequencies

Amplitude Leakage

Note: $T_1 = \frac{1}{100} = 0.01$ s.

or $f_m = 100$ Hz



Need:

$$N\Delta t = \frac{N}{f_s} = mT_1$$

Leakage

If the signal is non-periodic or nondeterministic waveform, we can control the accuracy of the spectrum amplitudes by varying DFT frequency resolution or total sampling time to minimize the leakage.

Summary I

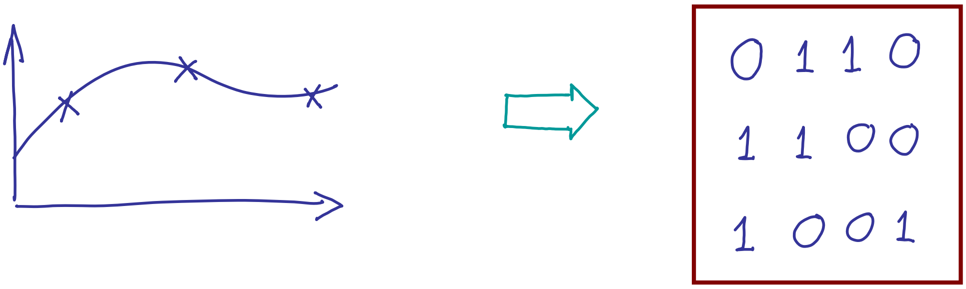
- To sampling any periodic waveform, choose both number of sampling points *N* and sampling rate f_s such that
  - the sampling theorem is satisfied
  - total sampling time is integer multiple of the fundamental frequency
- What to do for unknown signal
  - Sampling at different rate and compare the results. If the peak-frequency shifted, aliasing error occurs. Then try to increase sampling rate until peaks in spectrum is unchanged.
  - Avoid aliasing errors by filtering before processing FFT.

Outline II

- Digital Data Acquisition and its function
- Data Sampling
- Quantization: A/D, D/A converters

Quantization

After sampling the data, successive points are grabbed and then changed (quantized) to the digital (binary) form.



Binary

For instant, consider a 4-bit binary form:

– Number of binary (bins) is $2^4 = 16$

– Coding 1101 $\Rightarrow 1\times2^3 + 1\times2^2 + 0\times2^1 + 1\times2^0 = 13$

In general:

# of bits (N)	# of bins (2^N)
4	16
8	256
12	4096
14	16384
16	65536

Resolution of digitization

of Bins indicates the resolution, i.e.

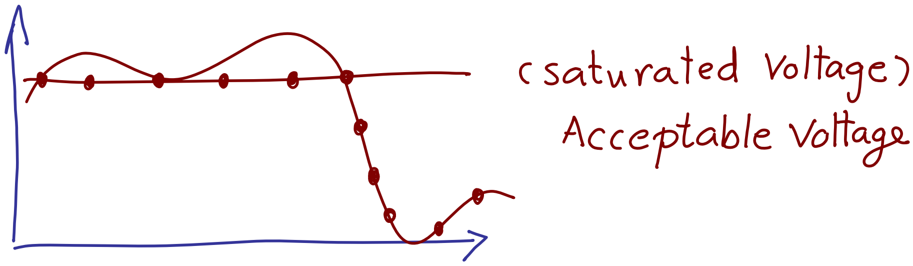
$$\text{resolution} = \frac{V_{max} - V_{min}}{(2^N - 1)} = \frac{\text{Range}}{[\#\text{ of Bins} - 1]}$$

$$\text{Quantization error} = \pm \frac{1}{2} \frac{(V_{max} - V_{min})}{(2^N - 1)}$$

Ex: Resolution of 4 bit binary for coding the analog signal with a range of -5 to +5 V (10 V span) is

$$\frac{10}{2^4 - 1} = \frac{10}{15} = 0.667 \text{ V}$$

Saturation error (clipping error)

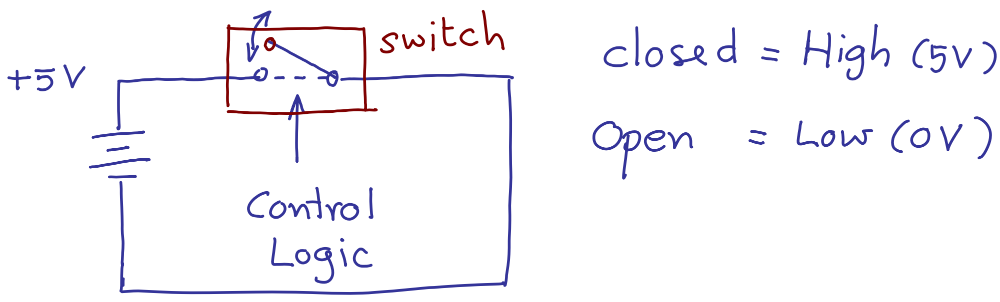


Digital 0-1 vs. Low-High signals

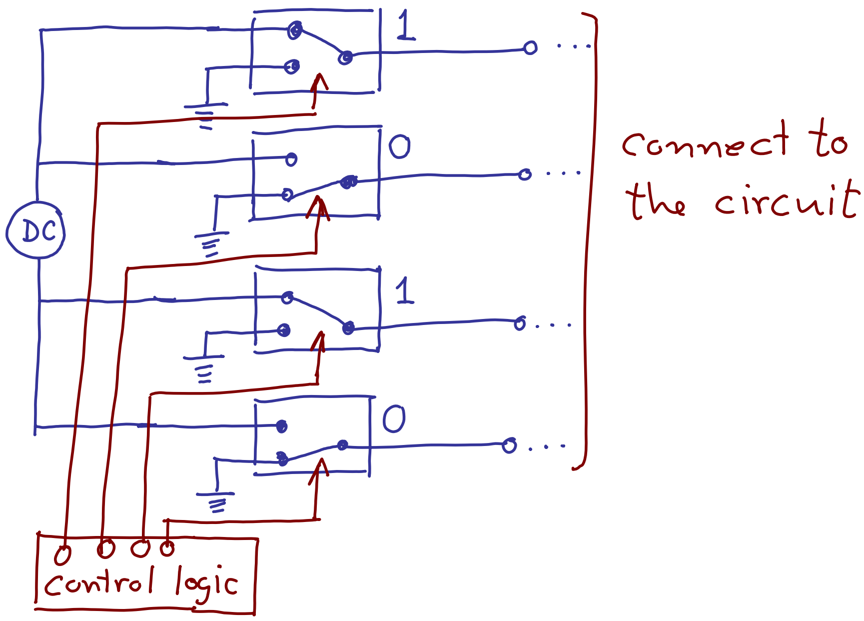
In digital circuit, binary number are formed using a combination of high and low voltages.

0 ⇒ Low and 1 ⇒ High

Open or closed switches connected in parallel are used as simple digital element to produce a binary code.



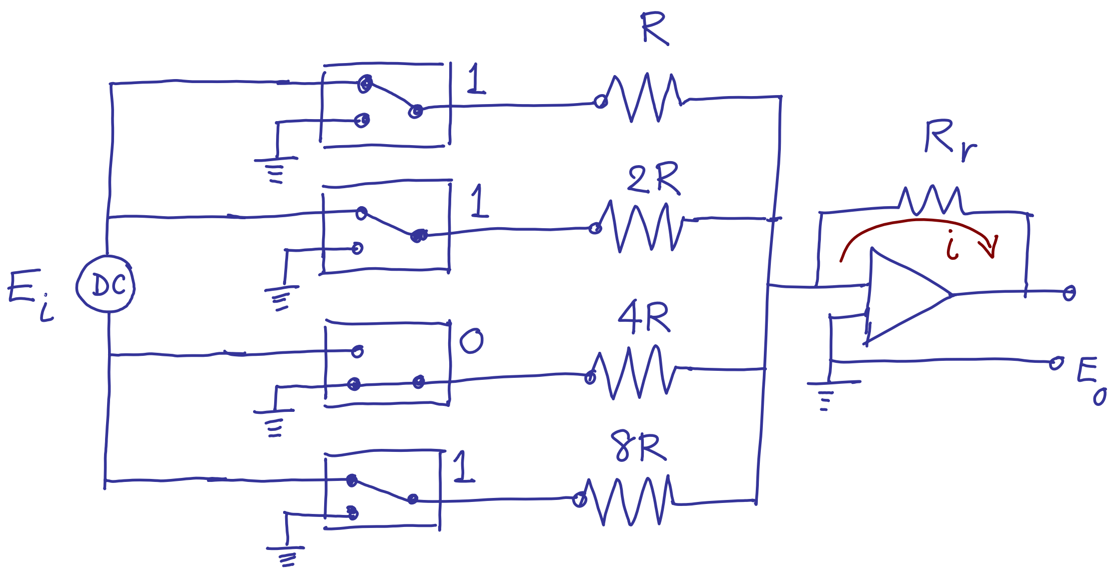
Simple on/off (1 or 0) switches



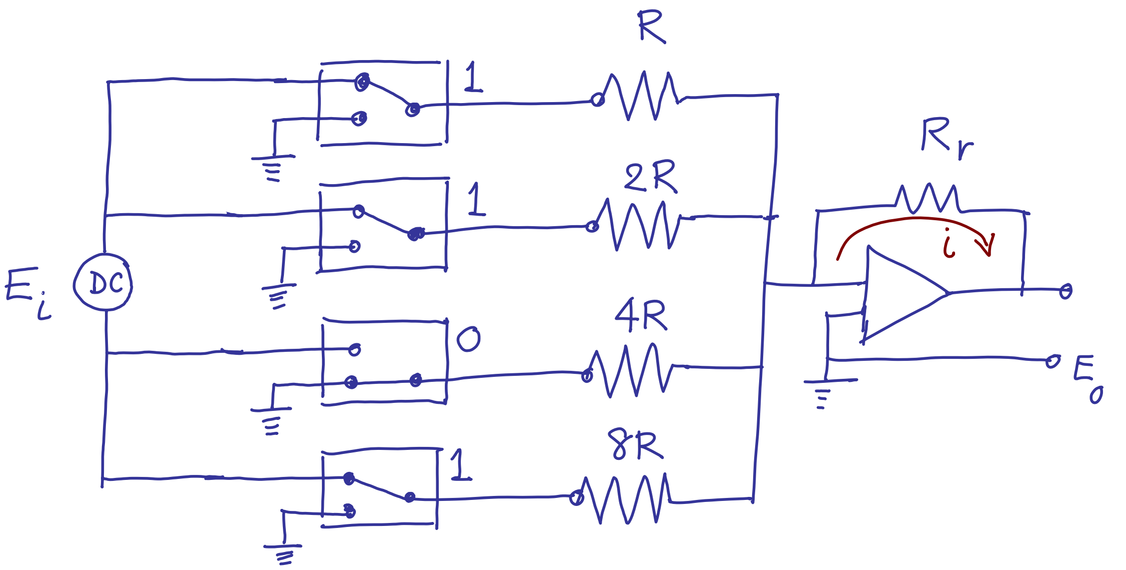
4 bits register transmitting 1010 in parallel

Digital-to-Analog (D/A) converter

Performed by adding up the contribution of each bit with a weight resistor into the op-amp.



Digital-to-Analog (D/A) converter



$$E_o = i R_r$$

where $$i = E_i \left(\frac{1}{R} + \frac{1}{2R} + \frac{0}{4R} + \frac{1}{8R} \right), \text{ for } 1101$$

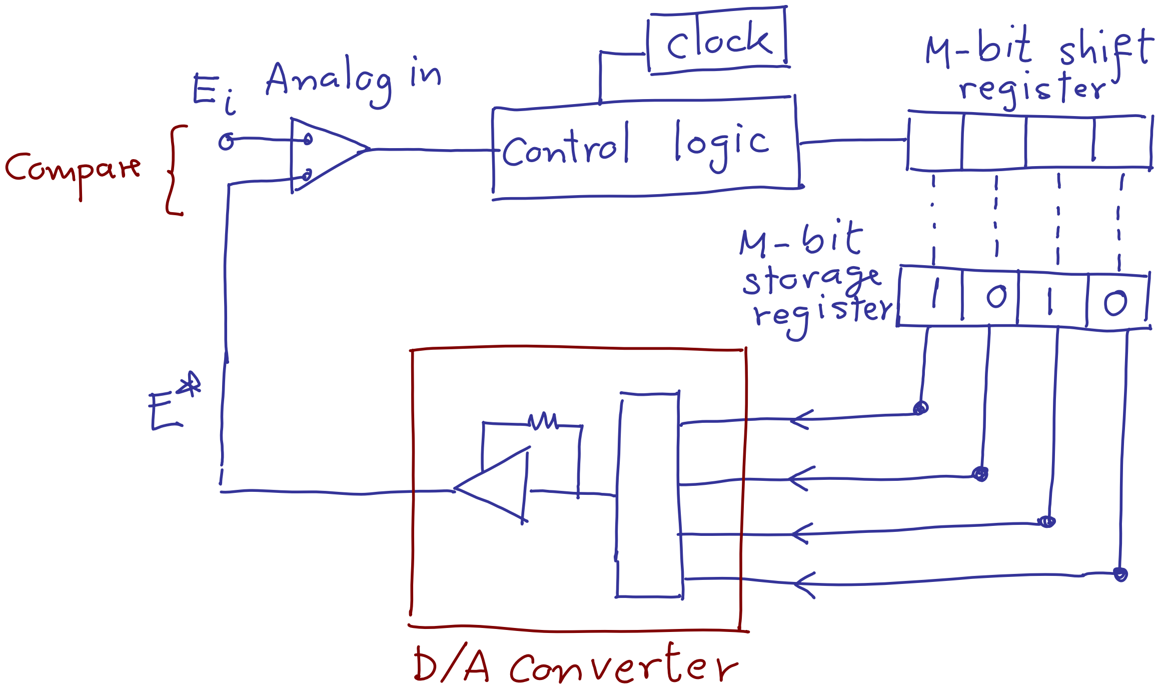
Analog-to-Digital (A/D) converter

Two basic kinds:

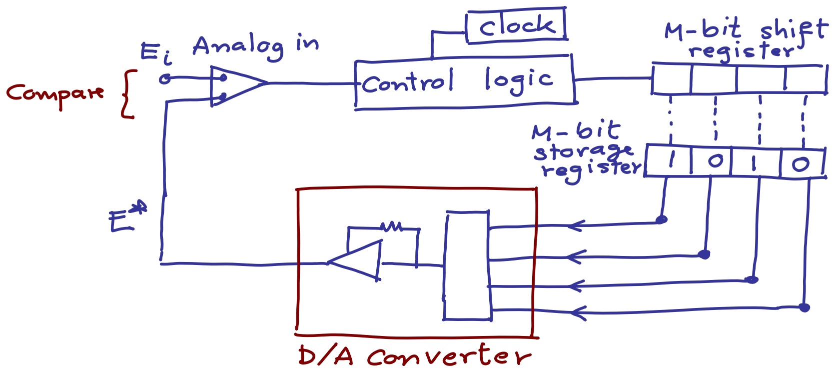
1. **Successive approximation converter** (the most used configuration) uses D/A generated voltages or known voltages to compare with the reference ones.

2. **Ramp converter**

A/D circuit (Successive Type)



Procedure

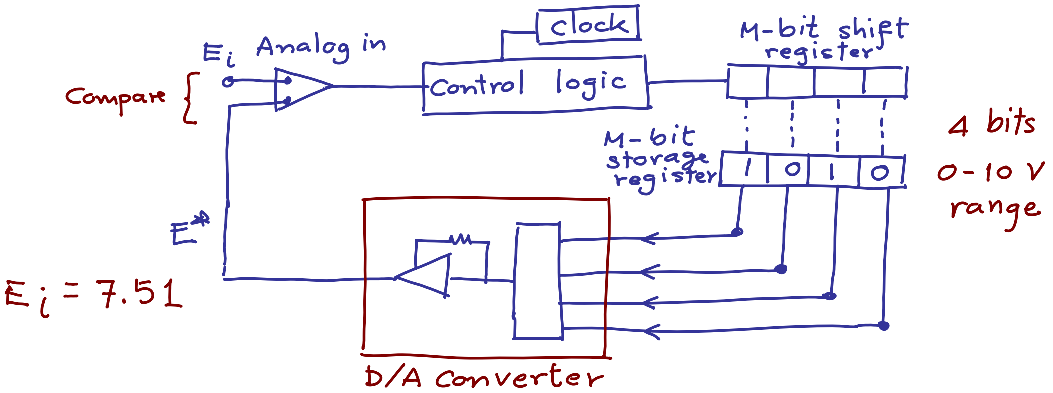


4 bits

0 - 10 V range

1) 1^{st} binary of 4 bits is set to '1' → $\underline{1}000$
2) E^* from D/A is compared with E_i
   - If $E^* > E_i$ bin code is reset to $\underline{0}000$
   - If $E^* < E_i$ bin code is kept as $\underline{1}000$
3) 2^{nd} binary is then set to '1' → $\times\underline{1}00$, repeat the process
4) Continue the process until the last digit

Procedure



Bin Code	E^*	$E^* < E_i$
1000	5.333	Y
1100	8	N
1010	6.667	Y
1011	7.333	Y
1011		

Noise in Measurement System

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Measurement and Instrumentations

Electrical Noise in Measurement System

Type	Cause	Correction
1) Thermal drift	Gradual changes in amplifier offset voltages and bias currents due to temperature, time and line voltage	• Choose consistent circuit materials • Select appropriate solder type • Minimize temperature gradients in circuit • Allow equipment the sufficient warm-up time and keep it away from significant hot or cold air source
2) Magnetic induced (induction) noise	Change in magnetic field near the circuit leads produces interfering voltages	• Remove the equipments such as power lines, motors, transformers, fluorescent lamps, relays, etc. from the sensitive signal circuits • Magnetic noise is minimized by twisting the conductor wires to be the loop-area • Use of enclosing magnetic or conductive shield to decrease noise. *(Shield functions by capturing changes and drained noise off to a ground)* • Shielding magnetic noise requires thick (2.5 mm) shields of ferromagnetic metals.
3) Electrostatic noise	Capacitance that present between adjacent conductors results in noise voltage coupled to the signal circuit electrostatically.	

Electrical Noise in Measurement System

Type	Cause	Correction
4) Ground-loop noise	Error voltage occurs when numerous instruments are grounded at different points causing difference in potentials and hence significant noise current adding to the circuit	• Ground all equipments at one point • Use isolate power sources and then find a good earth ground point for the entire system

1) $\Rightarrow$ Internal noise

2) – 4) $\Rightarrow$ External noise causing error voltage at power-line frequency

General ways to reduce electrical noise

- Keep signal wire short and away from electrical machinery
- Use twisted pair wires
- Use coaxial cable to reduce electric or magnetic fields but cost $$
- Shield cable and ground
- Integrate signal to smooth out noise
- Filtering