Data Acquisition for Embedded Systems

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Overview

- Signal Processing Basics
- Sampling
- Analog-to-Digital Conversion
- Digital-to-Analog Conversion

Signals and Systems

- Signal: a set of information and data
  - e.g. telephone, television signal, daily stock quote
  - Continuous and Discrete-time
  - Analog and Digital
- System: an entity that processes a set of signals (inputs) to yield another set of signals (outputs)
  - Characteristics: inputs, outputs, rules of operation
  - Classification:
    - Linear and non-linear
    - Constant parameter and time-varying parameter

(Trigonometric) Fourier Series

For a periodic function, $f(t)$, with period, $T = 1/f_0$,

$$f(t) = a_0 + \sum_{k=1}^{\infty} a_k \cos(2p f_0 kt) + \sum_{k=1}^{\infty} b_k \sin(2p f_0 kt)$$

$$= C_0 + \sum_{k=1}^{\infty} C_k \cos(2p f_0 kt + q_k)$$

f_0 = Fundamental frequency , $n \cdot f_0 = n^{th}$ harmonic of f_0

a_k, b_k = Amplitude of various harmonics

$$a_0 = C_0 = \frac{1}{T}\int_T f(t)\,dt$$

$$a_k = \frac{2}{T}\int_T f(t)\cos(2p f_0 kt)\,dt, \quad b_k = \frac{2}{T}\int_T f(t)\sin(2p f_0 kt)\,dt$$

$$C_n = \sqrt{a_k^2 + b_k^2},$$

Fourier Transform

- **Even if a function is not periodic, it can be described as a linear combination of an infinite number of orthogonal functions (In case of Fourier Transform, sinusoids). i.e. spectrum consists of a continuum of frequencies.**

  ($T \to \infty (f_0 \to 0)$ **)**
- **This spectrum can be defined by Fourier transform.**
- **For a signal *x(t)* with a spectrum *X(f)*, the followings hold:**

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2pft} dt \quad \text{(Fourier Transform)}$$

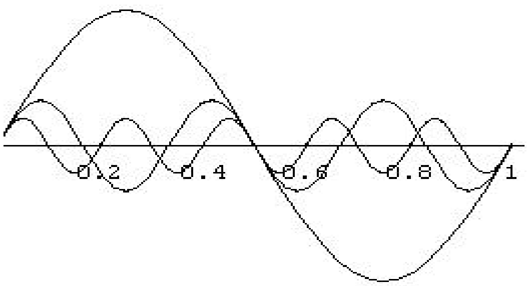
- **<x(t),X(f)>: Fourier transform pairs**

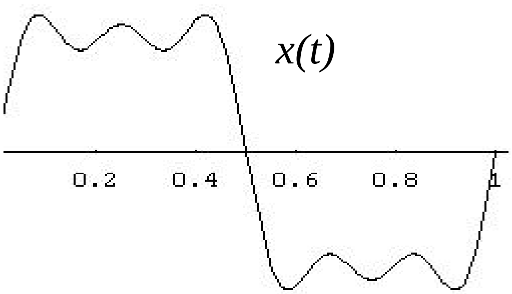
$$x(t) = \frac{1}{2p} \int_{-\infty}^{\infty} X(f) e^{j2pft} df \quad \text{(Inverse Fourier Transform}$$

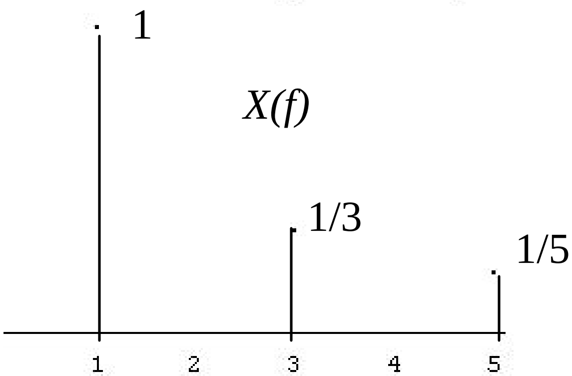
Time vs. Frequency Domain Representation

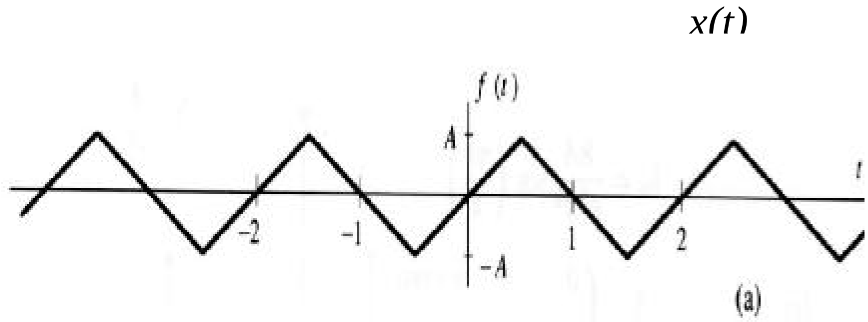
$$n(2pt) + \frac{1}{3}\sin(6pt) + \frac{1}{5}\sin(10pt)$$

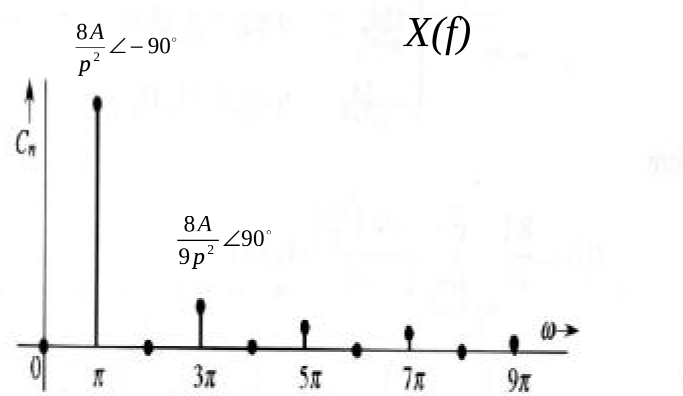
$$f(t) = \frac{8A}{p^2}\ \cos(pt - 90^\circ) + \frac{1}{9}\cos(3pt + 90^\circ) + \frac{1}{25}\cos(5pt -$$









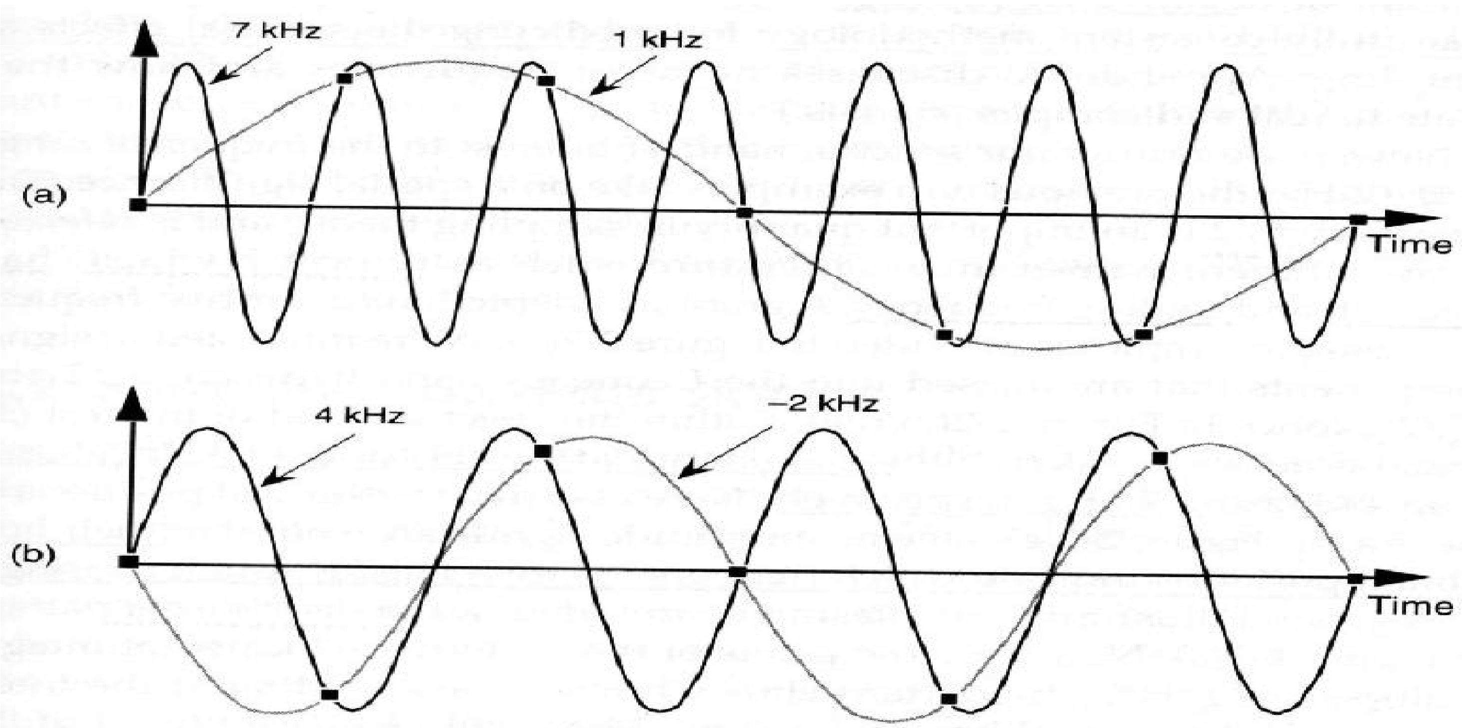


Sampling

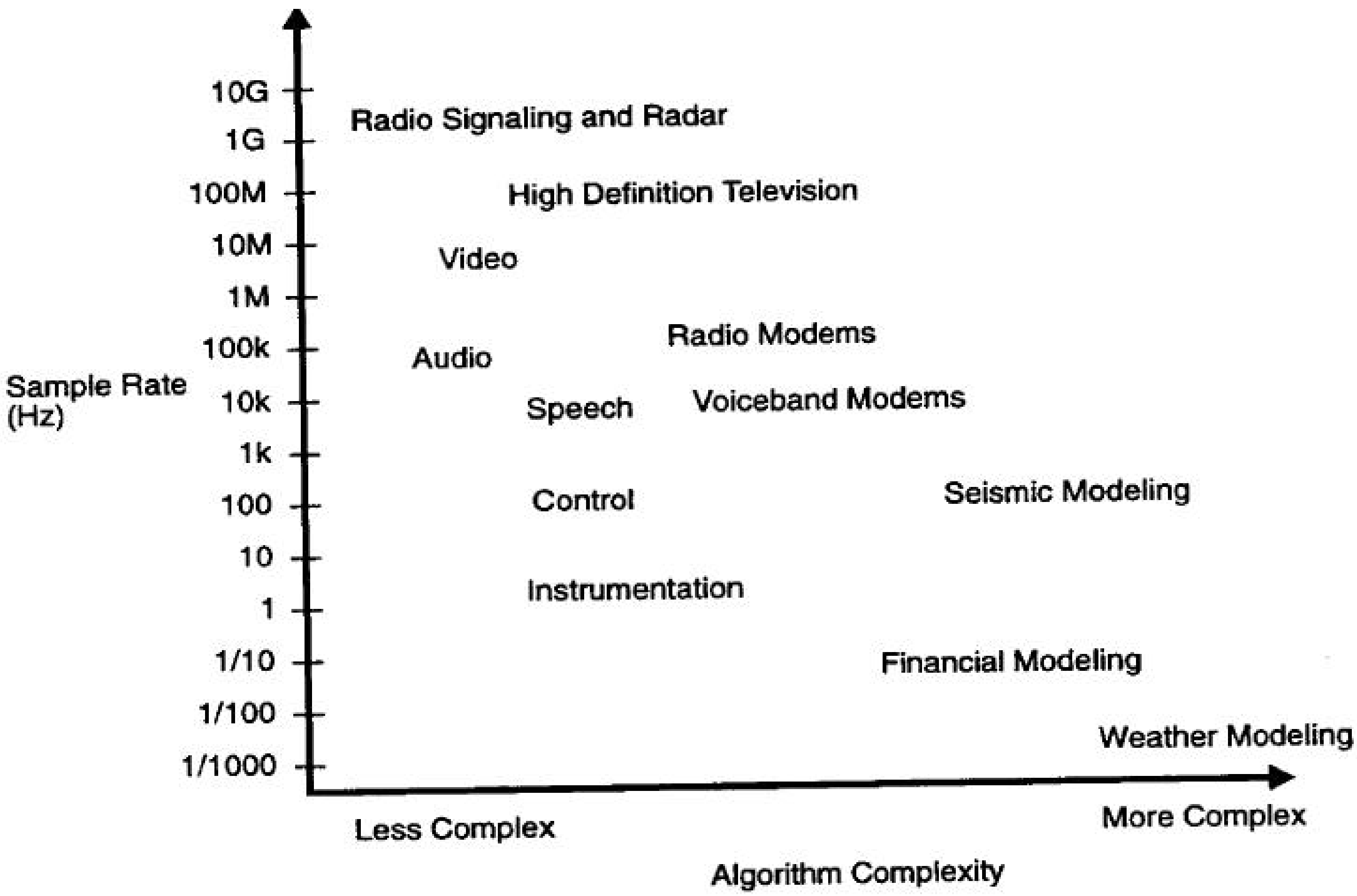
- **Sampling is a method of converting a continuous signal to a discrete set of values (samples)**
  - **should accurately represent the original signal**
  - **reduce the amount of information to be processed**
- **Major Concern:**
  - **How fast a given continuous signal must be sample to accurately represent the original**
- **Aliasing:**
  - **Frequency ambiguity [f_0 vs. (f_0+k f_s), k=integer]**

Sampling

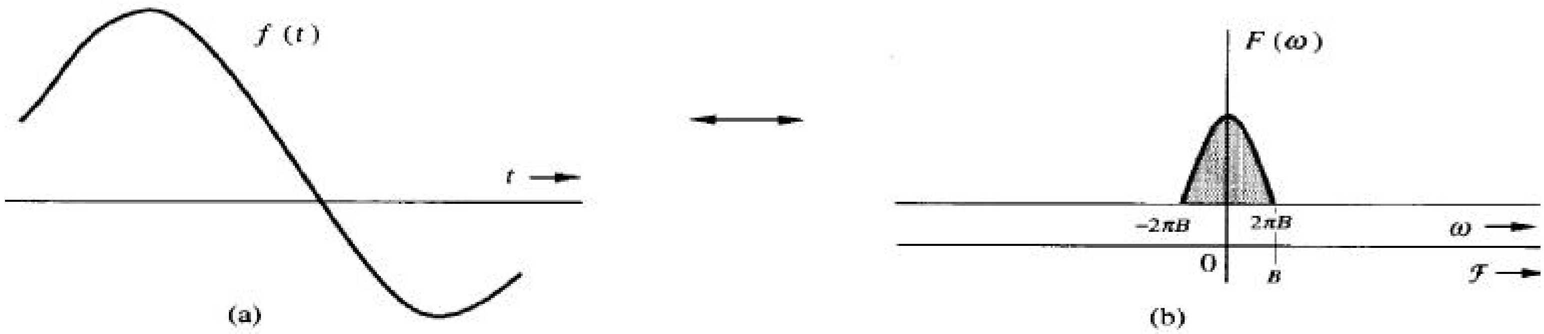
- **Examples below:**
  - **1) samples a slow (1KHz) signal with a 7KHz sine signal**
  - **2) samples a -2KHz signal with a 4KHz sine signal**



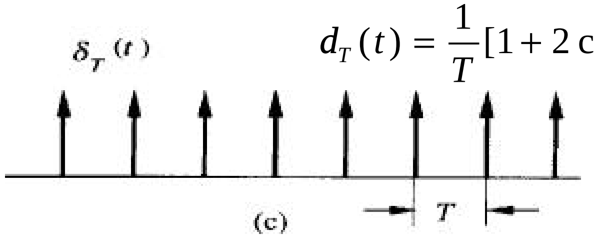
Typical Sampling Rate



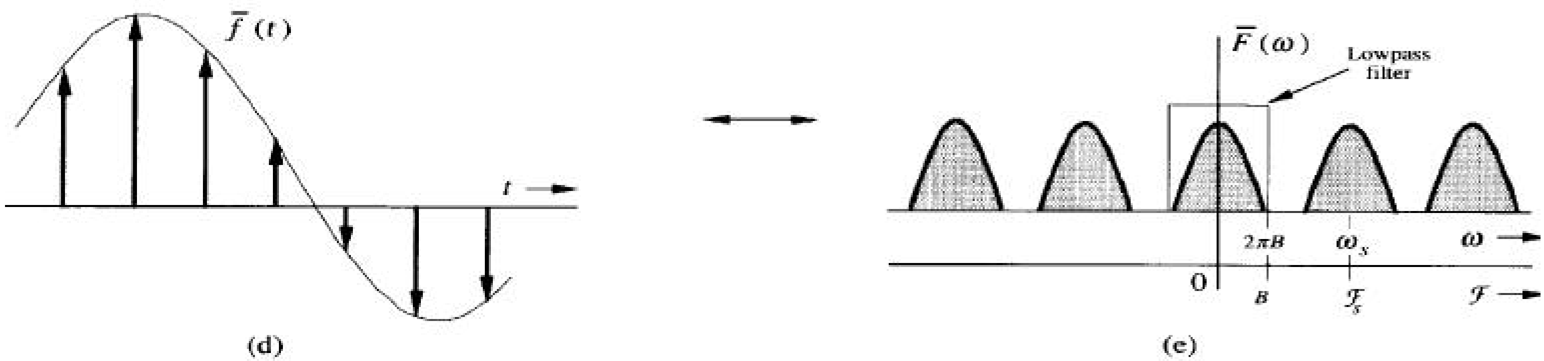
Sampled Signal & Fourier Spectrum



$$d_T(t) = \frac{1}{T}[1 + 2\cos 2p\, F_s t + 2\cos 4p\, F_s t + 2\cos 6p\, F_s t + \cdots]$$

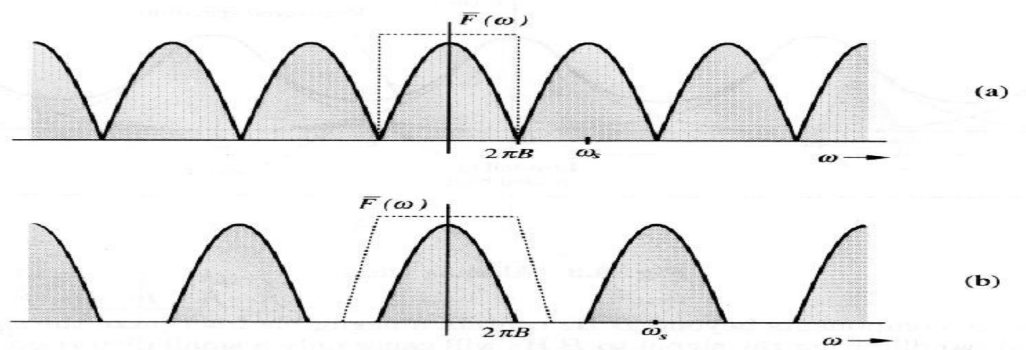


$$\overline{F}(w) = \frac{1}{T}\sum_{n=-\infty}^{\infty} F(w - nw_s)$$



Nyquist Rate

- **A signal which is (spectrum) band-limited to *B* Hz can be reconstructed exactly from its samples taken uniformly at a rate *R>2B* samples per second.**
  - **This is called** Nyquist rate **for the signal.**
- **Practical Difficulty**
  - **Low-pass filter limitation**
  - **Spectra should consist of repetitions of *F(w)* with a finite gap between successive cycles**

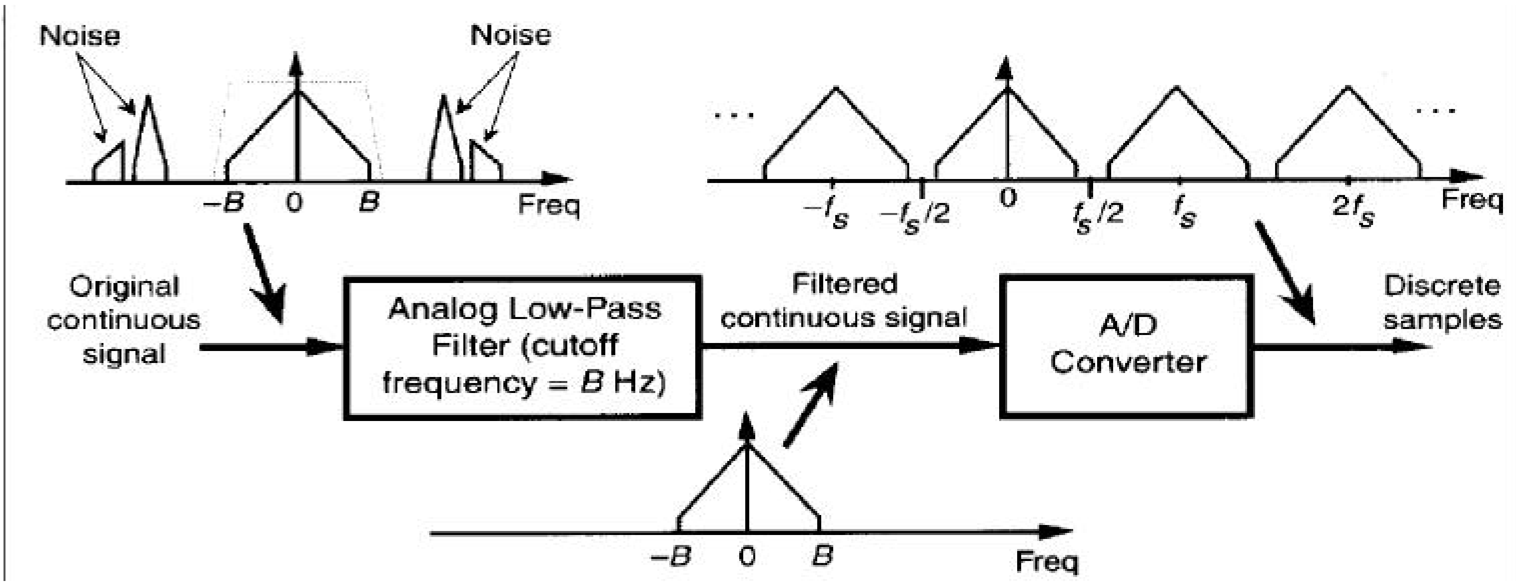


Aliasing Effect

- **All practical signals are time-limited**
  - **(non-band-limited)**
- **The spectrum $\overline{F}(w)$ consists of overlapping cycles of $F(w)$ repeating every F_s Hz**
  - **No longer complete recovery**
- **Aliasing (Spectral folding)**
  - **Folding frequency = 1/2 sampling frequency**

Elimination of Aliasing: Anti-Aliasing Filter

- Band-limited signal: No alias if F_s>Nyq.Rate
- Non-band-limited signal:
  - Alias results regardless of sampling rate
- Aliasing can be eliminated by band-limiting a signal before sampling
- Anti-aliasing filter: low-pass filter of bandwidth B Hz (B: effective bandwidth)



Analog vs. Digital - Pro's

Analog

- High bandwidth
- High resolution
- Specific control functions are available as off-the-shelf ICs
- Analysis and design methods are well-known

Digital

- Programmable solution
- Less sensitive to environment
- Can implement advanced control algorithms
- Capable of self-tuning, adaptive control, and nonlinear control functions
- Communication capability

Analog vs. Digital - Con’s

Analog

- Temperature drift
- Component aging
- Sensitive to noise
- Hardware design
- Can implement simple designs only
- No communication capability

Digital

- Data converter is required.
- Analysis and design methods are more complex
- Sampling & quantization error
- Computation delay limits the system bandwidth

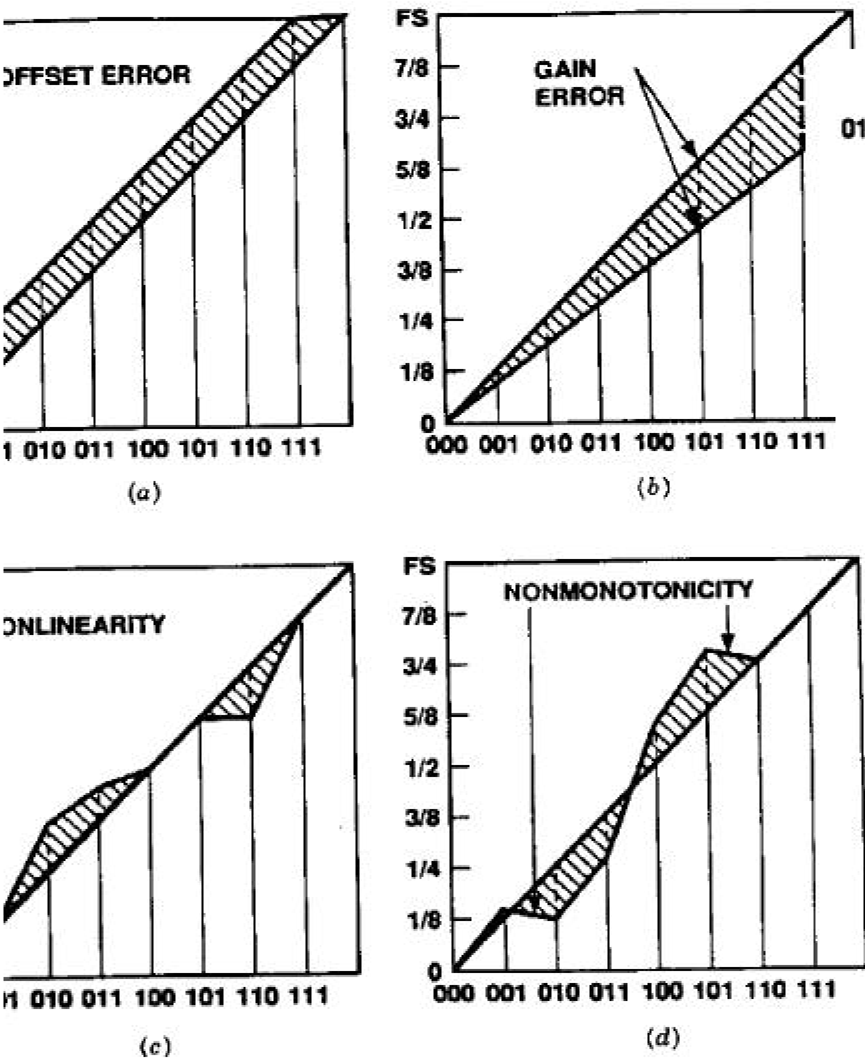
General Considerations

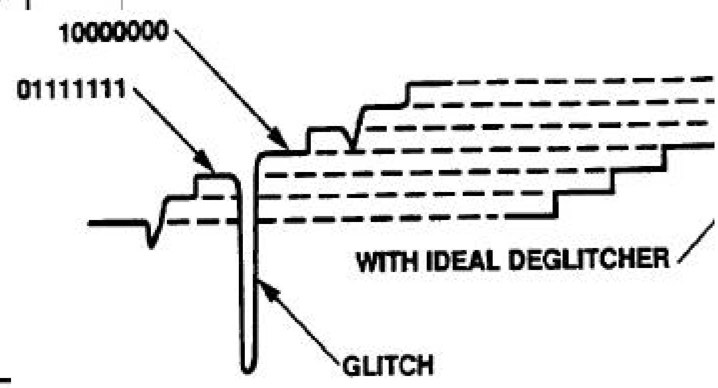
- Resolution: the number of bits in the input
- Figures of merit
  - resolution
  - temperature sensitivity
  - linearity
  - absolute accuracy (quantization error)
  - settling time, conversion time
  - price, complexity
- Errors
  - Static errors: Offset errors, Gain errors, Nonlinearity, Nonmonotonicity
  - Dynamic errors: degradation of S/N, glitches

Errors

static error

glitch





Analog to Digital and Digital to Analog Conversion

- Used in Embedded Systems to interact with physical world
- A-toD is the nexus of data acquisition process
  - ADC has been typically, the slowest, most complex, and/or most expensive single component in the data path
- Primary Concerns
  - Speed of Operation
  - Resolution (# of bits at the output)
  - Linearity, Non-monotonicity
  - Cost, Size, Power requirements, etc.

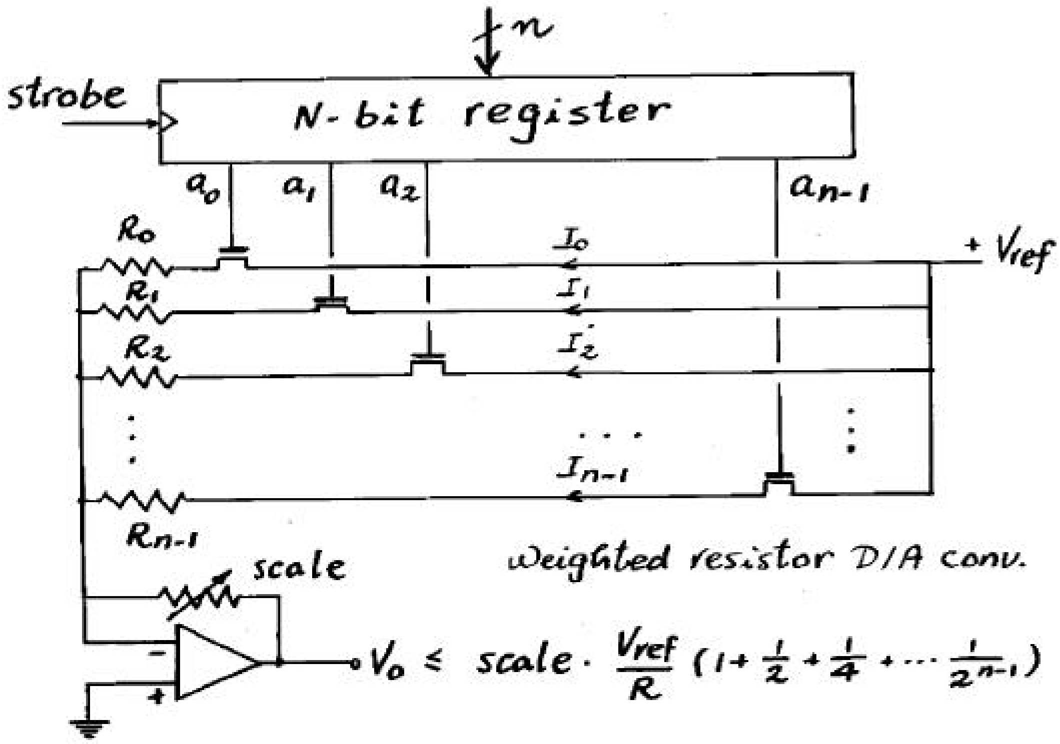
How fast should ADC be?

Applications	Appro. No of conversions per second	required conversion time
monitor and control	1 - 1000	15 - 1 ms
telephone voice	8,000	125 μs
CD-quality audio	85, 42.5, 21.3 K	50 - 12 μs
Video	$1-10 \times 10^6$ $3 \times 1-10 \times 10^6$	100 ns - 1 μs
radar	$100-1{,}000 \times 10^6$	1 - 10 ns

Common DAC Structures

- Weighted Resistor DAC
  - N resistors and switches (transistors)
- R/2R ladder DAC

Simple DAC



If the content of the register represents the number $\sum_{k=0}^{n-1} a_k 2^{-k}$,

$$R_0 = R,\ R_1 = 2R,\ R_2 = 4R \cdots R_{n-1} = 2^{n-1}R$$

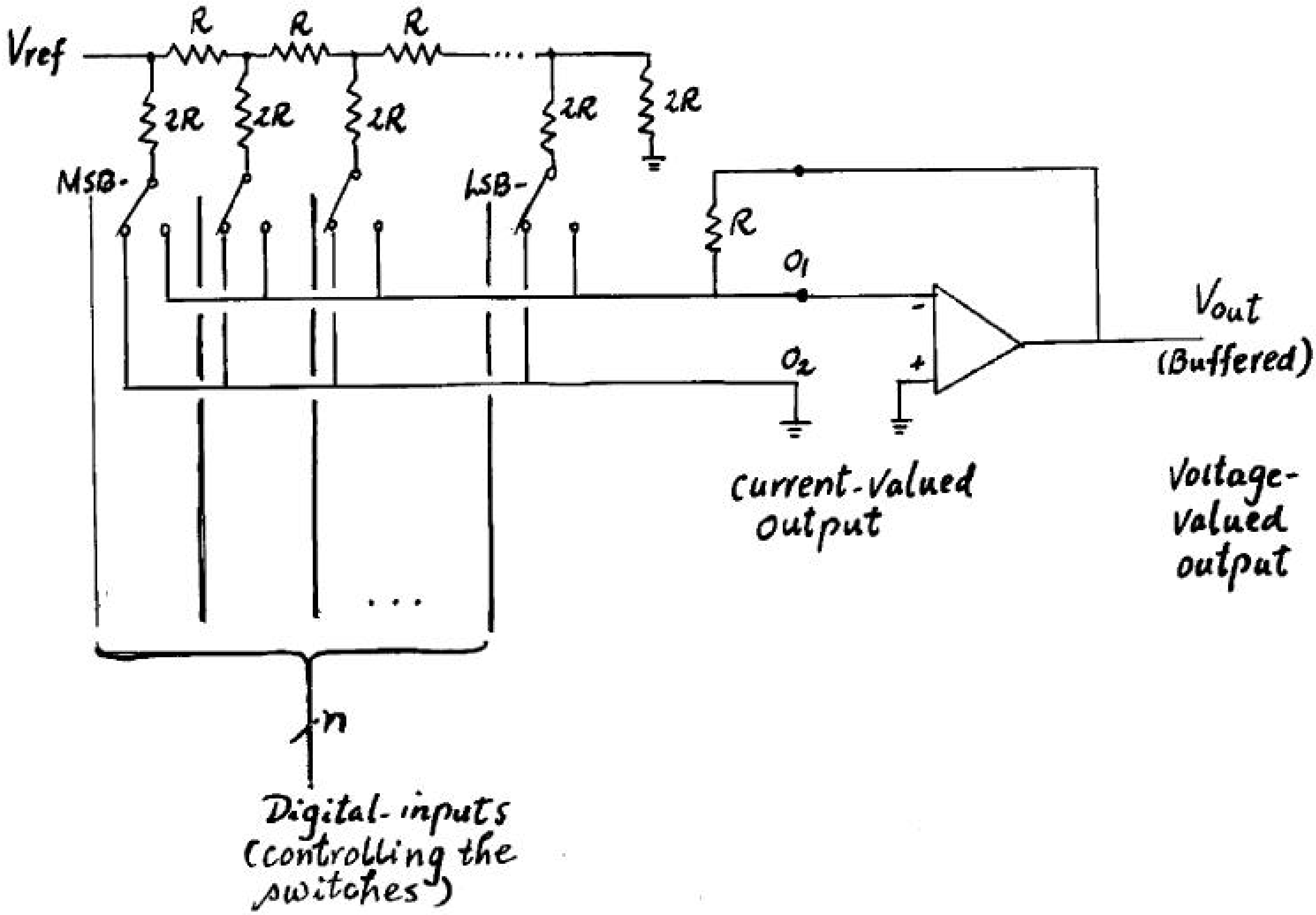
$$I_0 = I,\ I_1 = \frac{I}{2},\ I_2 = \frac{I}{4} \quad I_{n-1} = \frac{I}{2^{n-1}}$$

- Weighted resistor implementation
- Very stable V_{ref} required
- Straightforward in concept
- Impractical becaus a large resister rang is required
- Practically, can be upto 8bits

R/2R Ladder

- **A major improvement on the previous design**
  - **Use only two distinct resistor values**
  - **Output resistance is constant, independent of stages, *n***
  - **Fully modular - stages can be added or deleted w/o compromising the design**
- **Each digit *I* adds ~V/*I* volts to the output sum**
- **Rather slow**

R/2R Ladder DAC



Digital to Analog Conversion

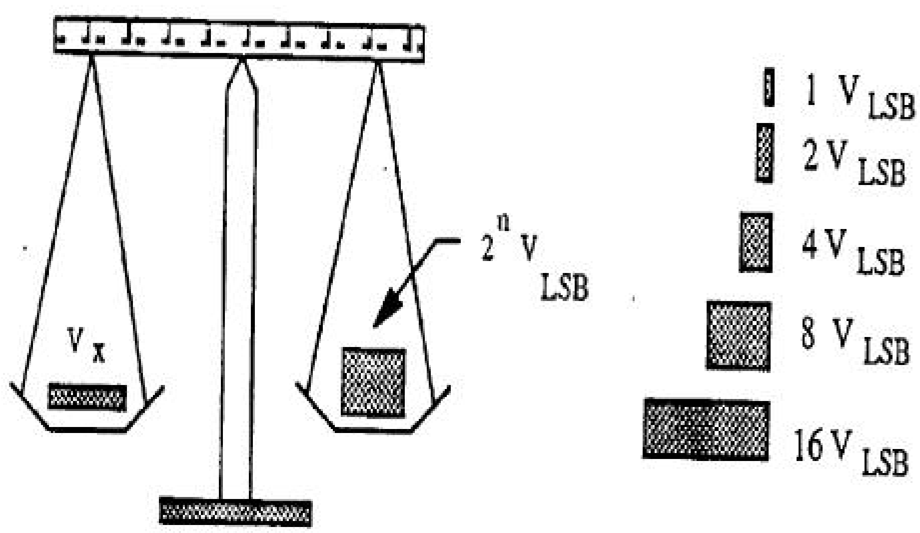
- **Convert each bit (digit) to a weighted voltage level, sum them**
- **Can use A-to-D to perform D-to-A.**

Common ADC Structures

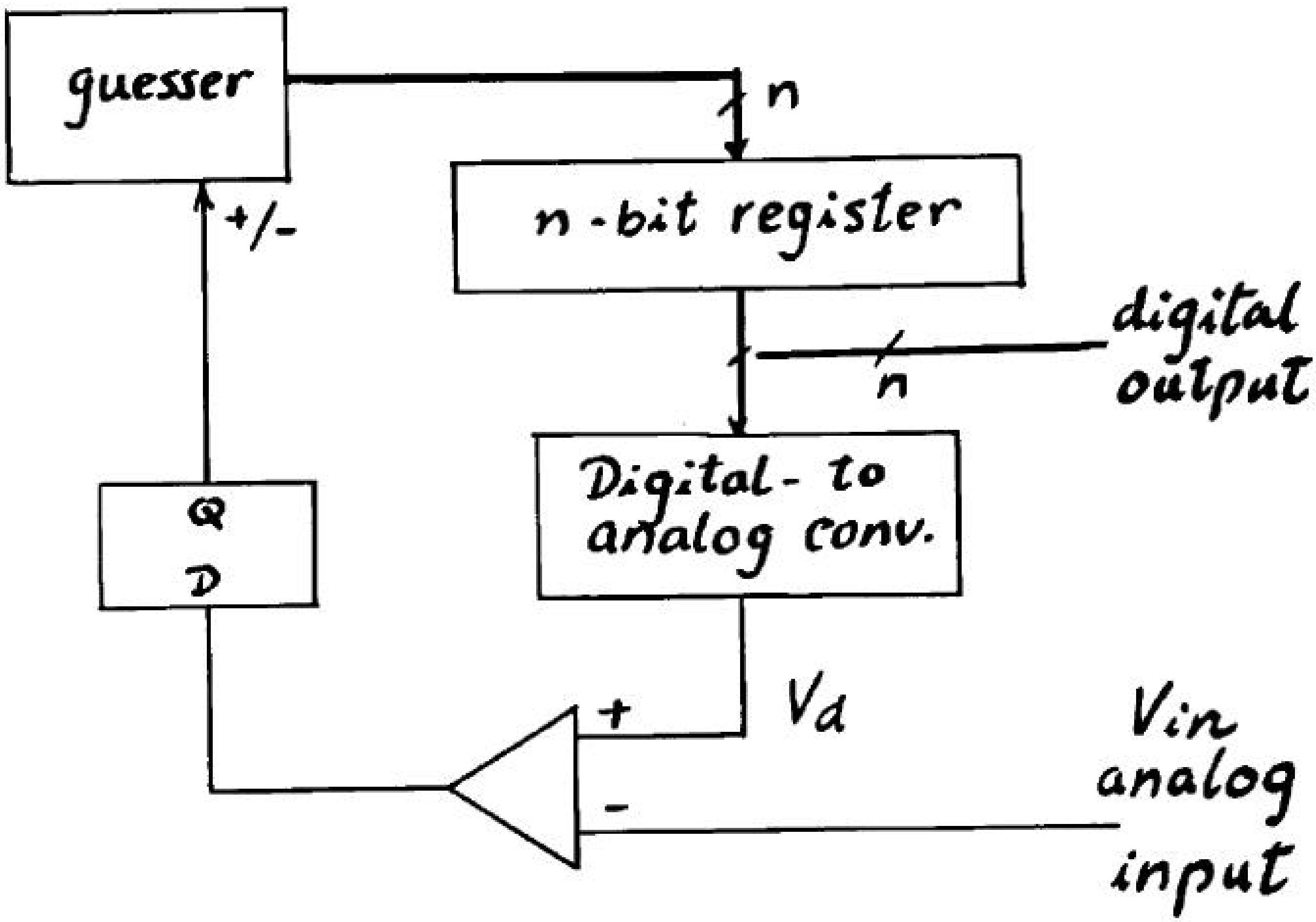
- **Successive-Approximation (serial) ADC - medium**
  - 12-16 bits
  - 1-10 micro seconds - slow
  - Various forms of approximation:
    - simple counter
    - initial guess and up/down count
    - digit by digit, starting at msb
- **Parallel (“Flash”) ADC - fast, but very expensive**
  - uses a resistive “divider” to generate reference voltage for each bit
  - <= 8 bits
  - fast

Successive Approximation

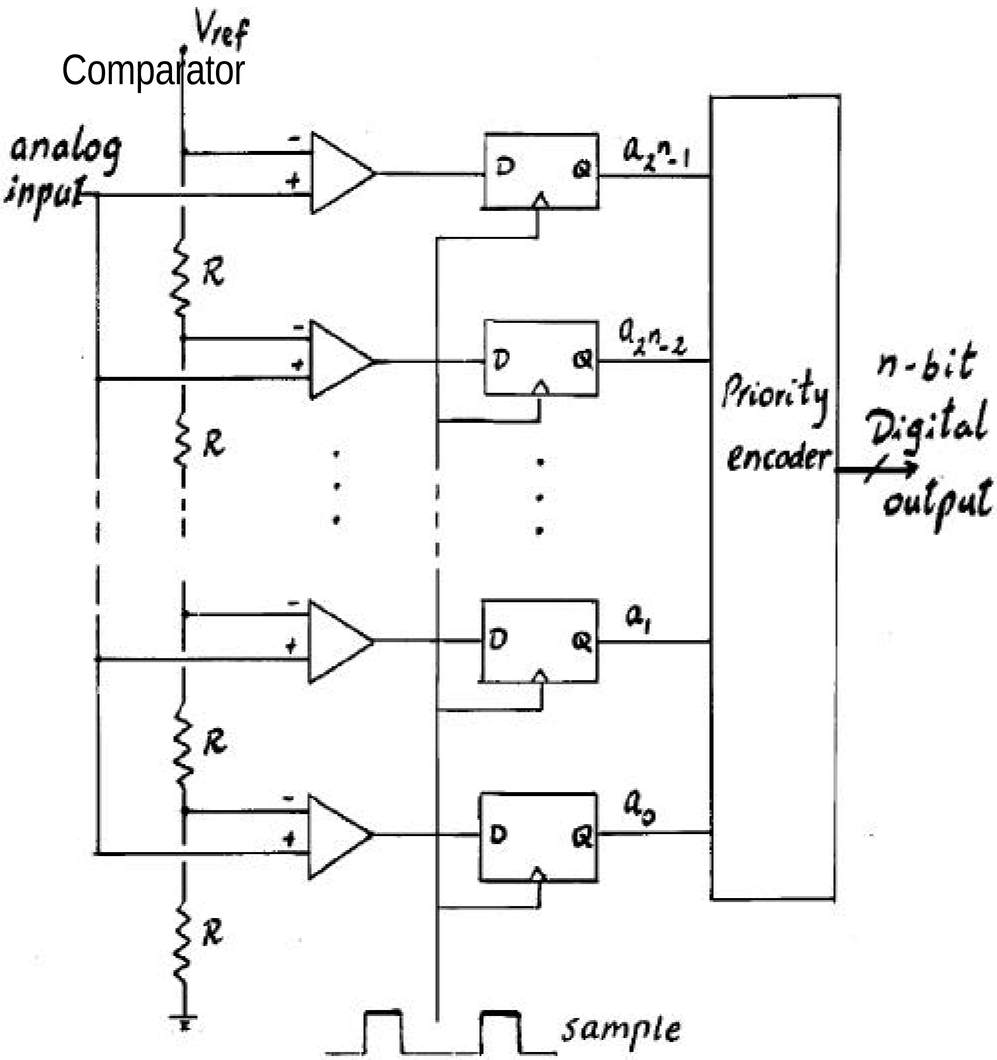
- Conversion algorithm
  - Start with an initial guess
  - Output the guess to DAC, compare the result of DAC to V_{in}
  - Improve the guess
- Analogy:



A-to-D via Counting or Successive Approximation



Flash(Parallel) ADC- fastest



- ›nversion speed: 4-50ns
- ›solution: limited to 8 bits
  ›e to chip area and dc
  ›wer dissipation
- ›mplexity: exponential to ›

Summary

- Sampling of analog signals and digital conversion
- Sampling Theorem
  - Nyquist rate
  - Aliasing
- Digital to Analog Conversion
  - R/2R
- Analog to Digital Conversion
  - Flash
  - Successive Approximation