

Digital Signal Processing

Module 7: Stochastic Signal Processing and Quantization

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- ▶ **Module 7.1:** Stochastic signals
- ▶ **Module 7.2:** Quantization
- ▶ **Module 7.2:** A/D and D/A conversion

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Module 7.1: Stochastic signal processing

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- ▶ A simple random signal

- ▶ Power spectral density

- ▶ Filtering a stochastic signal

- ▶ Noise

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- ▶ deterministic signals are known in advance: $x[n] = \sin(0.2 n)$
- ▶ interesting signals are *not* known in advance: $s[n] = \text{what I'm going to say next}$
- ▶ we usually know something, though: $s[n]$ is a speech signal
- ▶ stochastic signals can be described probabilistically
- ▶ can we do signal processing with random signals? Yes!
- ▶ will not develop stochastic signal processing rigorously but give enough intuition to deal with things such as “noise”

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For each new sample, toss a fair coin:

$$x[n] = \begin{cases} +1 & \text{if the outcome of the } n\text{-th toss is head} \\ -1 & \text{if the outcome of the } n\text{-th toss is tail} \end{cases}$$

- ▶ each sample is independent from all others
- ▶ each sample value has a 50% probability

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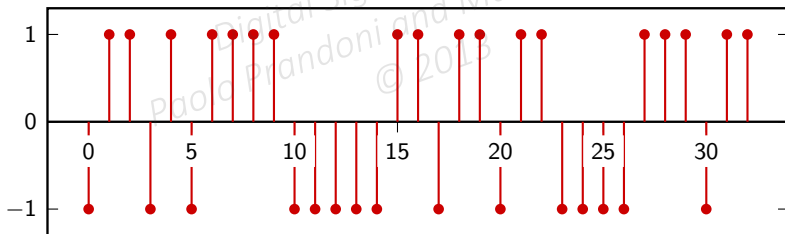
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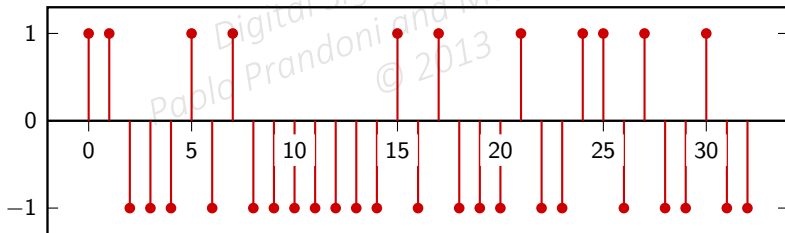
- ▶ every time we turn on the generator we obtain a different *realization* of the signal
- ▶ we know the “mechanism” behind each instance
- ▶ but how can we analyze a random signal?

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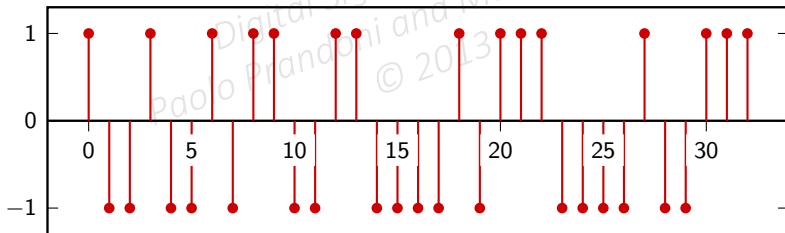
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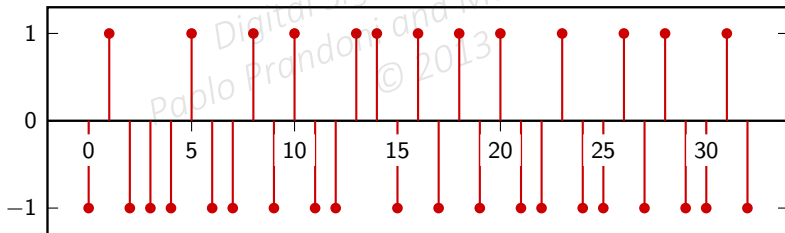
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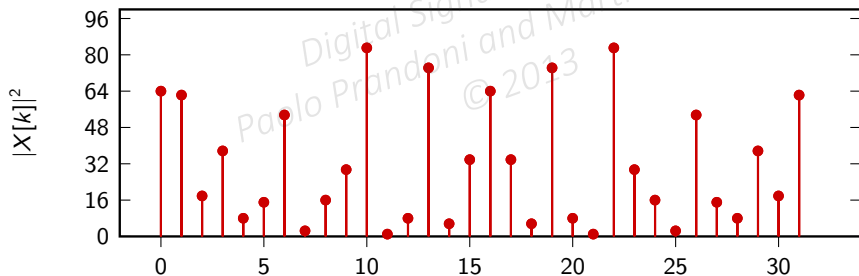
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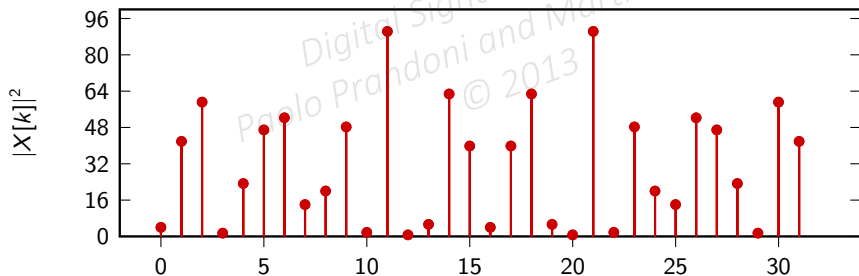
- ▶ let's try with the DFT of a finite set of random samples
- ▶ every time it's different; maybe with more data?
- ▶ no clear pattern... we need a new strategy

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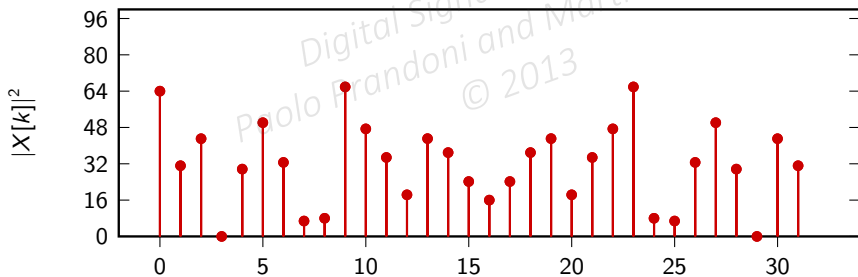
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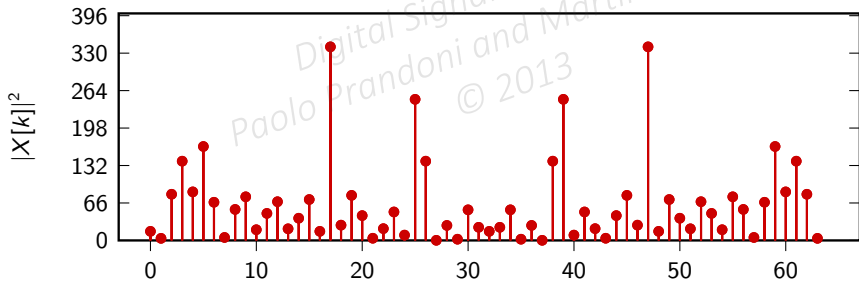
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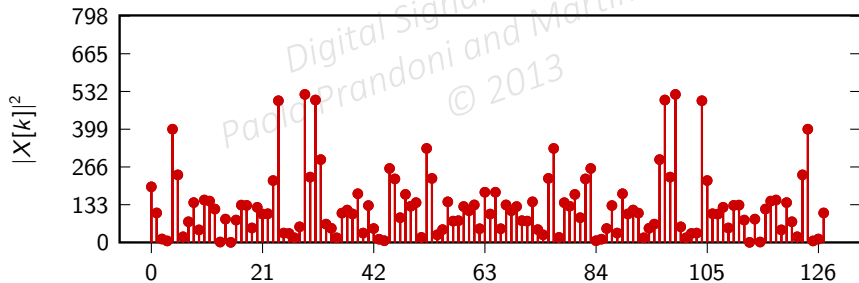
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- ▶ in probability theory the average is across realizations and it's called *expectation*
- ▶ for the coin-toss signal:

$$E[x[n]] = -1 \cdot P[\text{n-th toss is tail}] + 1 \cdot P[\text{n-th toss is head}] = 0$$

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- ▶ the coin-toss signal has infinite energy (see Module 2.1):

$$E_x = \lim_{N \rightarrow \infty} \sum_{n=-N}^N |x[n]|^2 = \lim_{N \rightarrow \infty} (2N + 1) = \infty$$

- ▶ however it has finite power over any interval:

$$P_x = \lim_{N \rightarrow \infty} \frac{1}{2N + 1} \sum_{n=-N}^N |x[n]|^2 = 1$$

let's try to average the DFT's square magnitude, normalized:

- ▶ pick an interval length N
- ▶ pick a number of iterations M
- ▶ run the signal generator M times and obtain M N -point realizations
- ▶ compute the DFT of each realization
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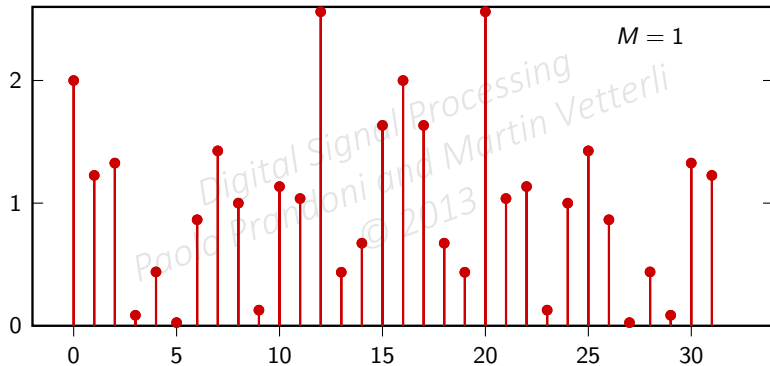
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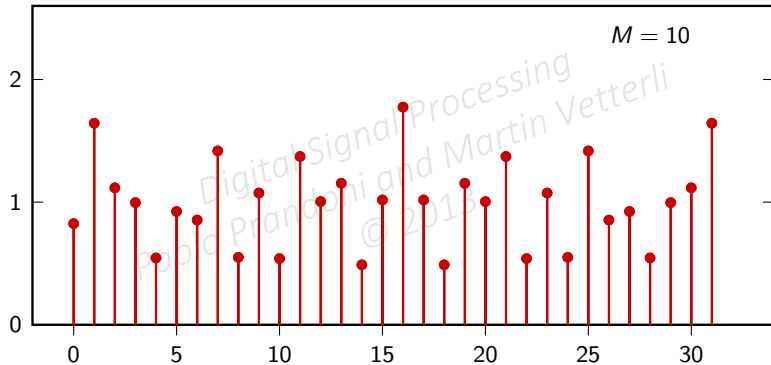
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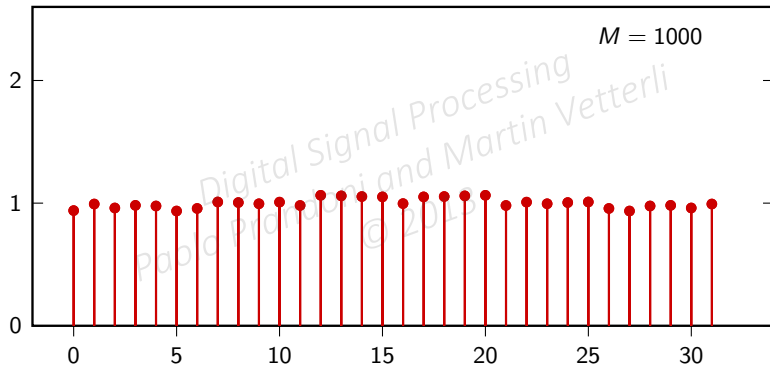
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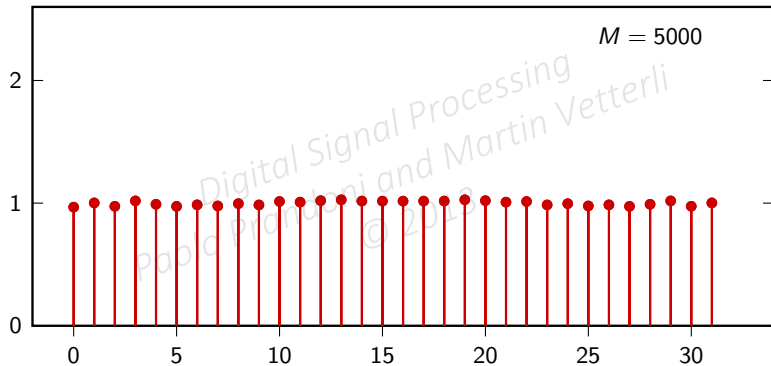
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- ▶ if $|X_N[k]|^2$ tends to the energy distribution in frequency...
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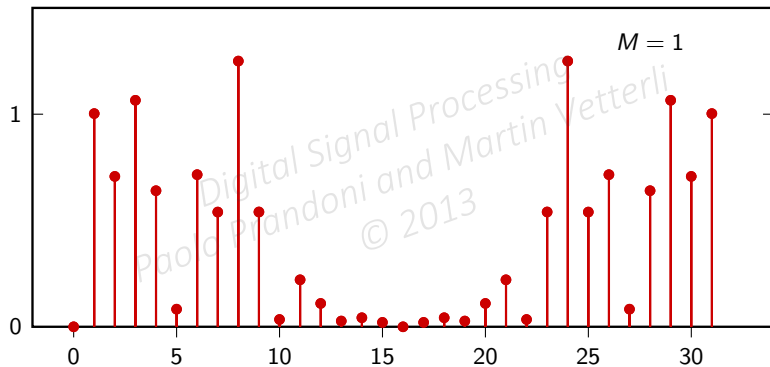
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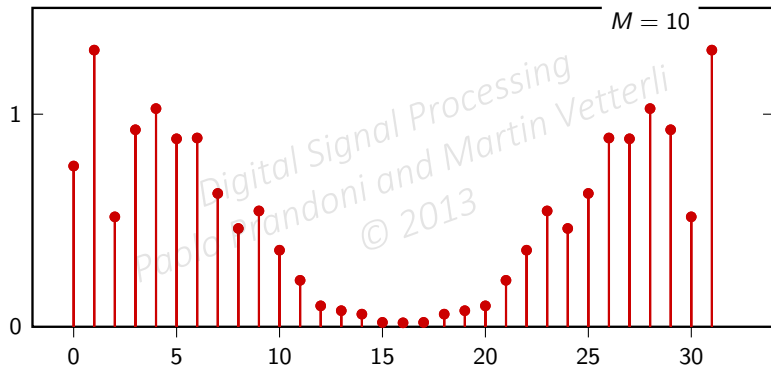
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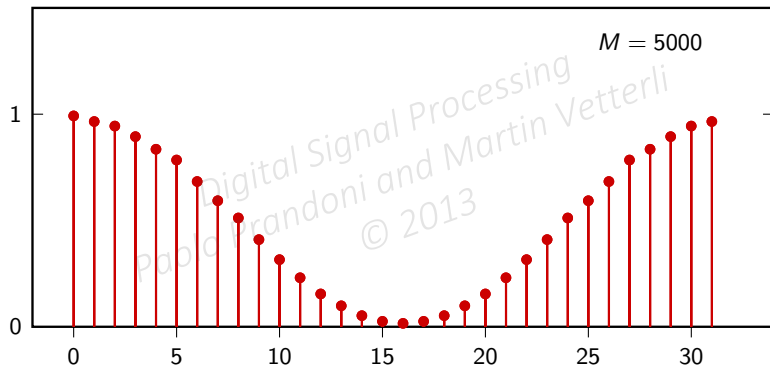
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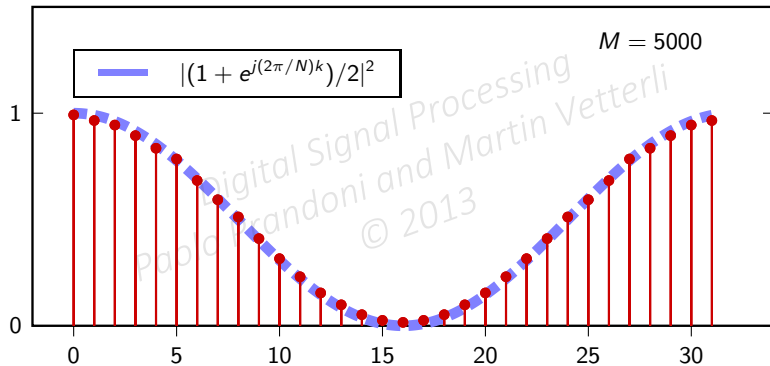
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- ▶ it looks like $P_y[k] = P_x[k] |H[k]|^2$, where $H[k] = \text{DFT} \{h[n]\}$
- ▶ can we generalize these results beyond a finite set of samples?

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- ▶ a stochastic process is characterized by its power spectral density (PSD)

- ▶ it can be shown (see the textbook) that the PSD is

where $r_x[n] = E[x[k]x[n-k]]$ is the autocorrelation of the process.

- ▶ for a filtered stochastic process $y[n] = \mathcal{H}\{x[n]\}$, it is:

$$P_y(e^{j\omega}) = |H(e^{j\omega})|^2 P_x(e^{j\omega})$$

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- thermal noise
- sum of extraneous interferences
- quantization and numerical errors
- ...

► we can model noise as a stochastic signal

► the most important noise is white noise

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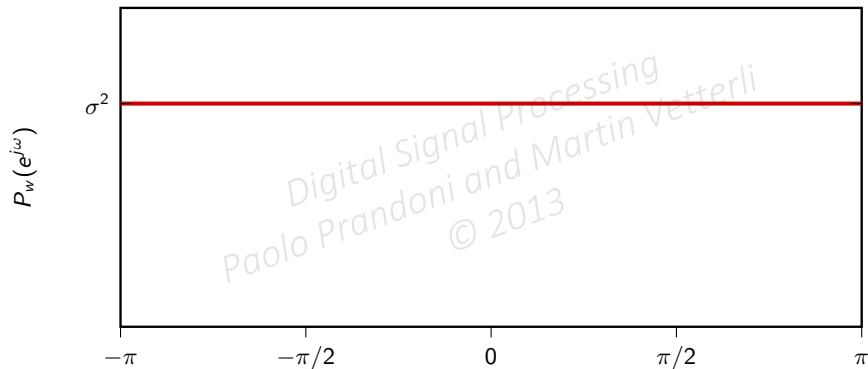
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- ▶ distribution is important to estimate bounds for the signal
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END OF MODULE 7.1

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Module 7.2: Quantization

- **Quantization**

- Uniform quantization and error analysis

- Clipping, saturation, companding

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- ▶ we need to map the range of a signal onto a finite set of values
- ▶ irreversible loss of information \rightarrow quantization noise

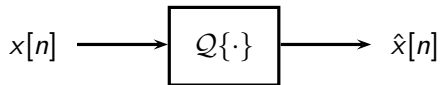
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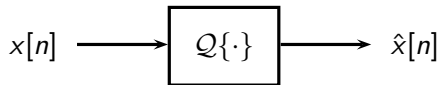
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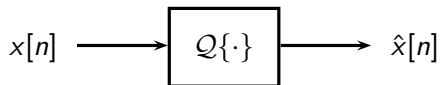
- ▶ storage budget (bits per sample)
- ▶ storage scheme (fixed point, floating point)
- ▶ properties of the input
 - range
 - probability distribution

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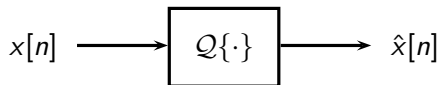
- ▶ storage budget (bits per sample)
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 - range
 - probability distribution



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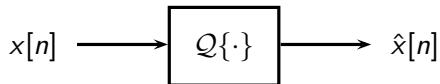
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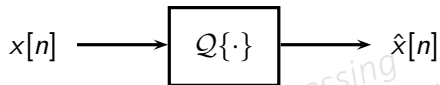
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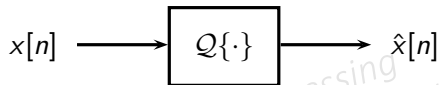
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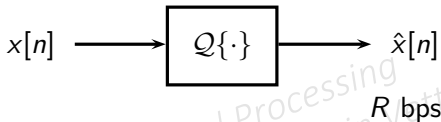
The simplest quantizer:

- ▶ each sample is encoded individually (hence *scalar*)
- ▶ each sample is quantized independently (memoryless quantization)
- ▶ each sample is encoded using R bits



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Assume input signal bounded: $A \leq x[n] \leq B$ for all n :

- ▶ each sample quantized over 2^R possible values $\Rightarrow 2^R$ intervals.
- ▶ each interval associated to a quantization value



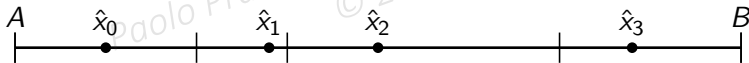
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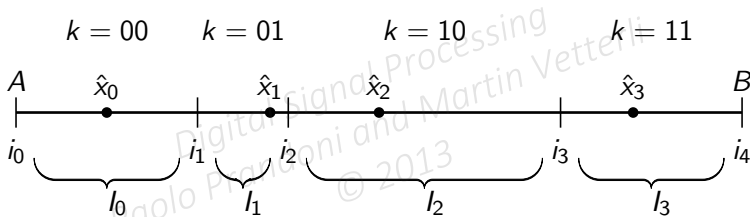


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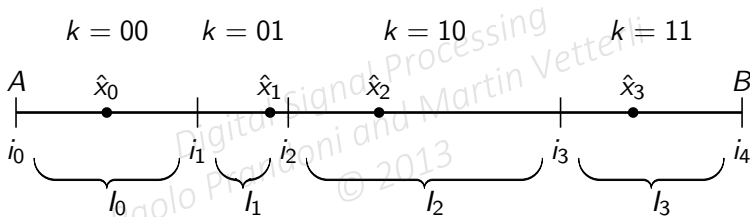


Example for $R = 2$:



- ▶ what are the optimal interval boundaries i_k ?
- ▶ what are the optimal quantization values \hat{x}_k ?

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$$e[n] = Q\{x[n]\} - x[n] = \hat{x}[n] - x[n]$$

- ▶ model $x[n]$ as a stochastic process
- ▶ model error as a white noise sequence:
 - error samples are uncorrelated
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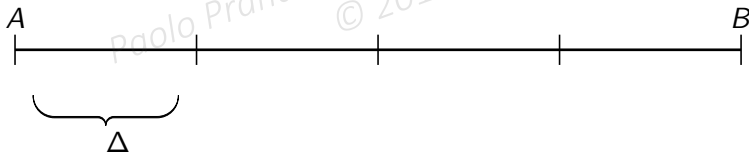
- ▶ simple but very general case
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Mean Square Error is the variance of the error signal:

$$\sigma_e^2 = E [|Q\{x[n]\} - x[n]|^2]$$

$$= \int_{-B}^B f_x(\tau) (Q\{\tau\} - \tau)^2 d\tau$$

$$= \sum_{k=0}^{2^R-1} \int_{I_k} f_x(\tau) (\hat{x}_k - \tau)^2 d\tau$$

error depends on the probability distribution of the input

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error depends on the probability distribution of the input

Uniform-input hypothesis:

$$f_x(\tau) = \frac{1}{B-A}$$

$$\sigma_e^2 = \sum_{k=0}^{2^R-1} \int_{I_k} \frac{(\hat{x}_k - \tau)^2}{B-A} d\tau$$

Let's find the optimal quantization point by minimizing the error

$$\begin{aligned}\frac{\partial \sigma_e^2}{\partial \hat{x}_m} &= \frac{\partial}{\partial \hat{x}_m} \sum_{k=0}^{2^R-1} \int_{I_k} \frac{(\hat{x}_k - \tau)^2}{B-A} d\tau \\ &= \frac{\partial}{\partial \hat{x}_m} \int_{I_m} \frac{2(\hat{x}_m - \tau)}{B-A} d\tau \\ &= \frac{(\hat{x}_m - \tau)^2}{B-A} \Big|_{A+m\Delta}^{A+m\Delta+\Delta}\end{aligned}$$

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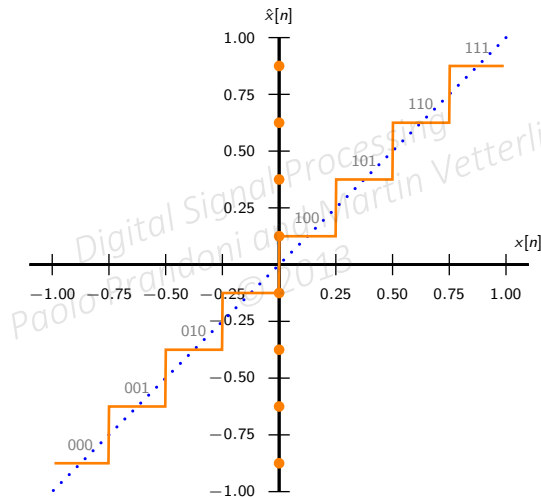
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Minimizing the error:

$$\frac{\partial \sigma_e^2}{\partial \hat{x}_m} = 0 \quad \text{for } \hat{x}_m = A + m\Delta + \frac{\Delta}{2}$$

optimal quantization point is the interval's midpoint, for all intervals

Uniform 3-Bit quantization function



Quantizer's mean square error:

$$\begin{aligned}\sigma_e^2 &= \sum_{k=0}^{2^R-1} \int_{A+k\Delta}^{A+k\Delta+\Delta} \frac{(A+k\Delta+\Delta/2-\tau)^2}{B-A} d\tau \\ &= 2^R \int_0^\Delta \frac{(\Delta/2-\tau)^2}{B-A} d\tau \\ &= \frac{\Delta^2}{12}\end{aligned}$$

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$$\sigma_e^2 = \Delta^2/12, \quad \Delta = (B - A)/2^R$$

- ▶ signal energy

- ▶ signal to noise ratio

- ▶ in dB

$$\text{SNR}_{\text{dB}} = 10 \log_{10} 2^{2R} \approx 6R \text{ dB}$$

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The “6dB/bit” rule of thumb



- ▶ a compact disk has 16 bits/sample:

$$\max \text{SNR} = 96\text{dB}$$

- ▶ a DVD has 24 bits/sample

$$\max \text{SNR} = 144\text{dB}$$

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The “6dB/bit” rule of thumb



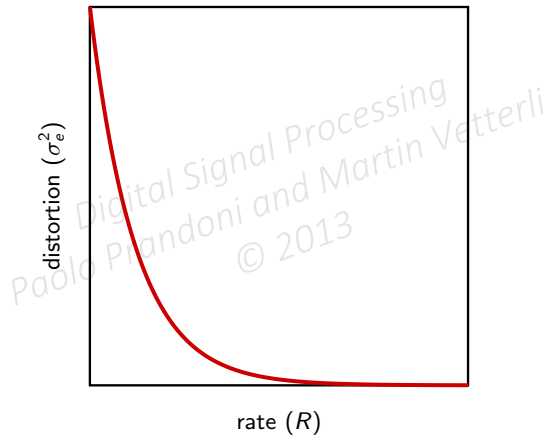
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If input is not bounded to $[A, B]$:

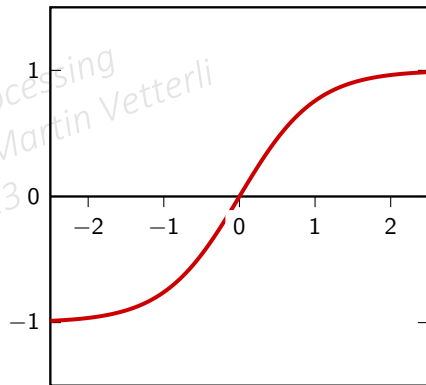
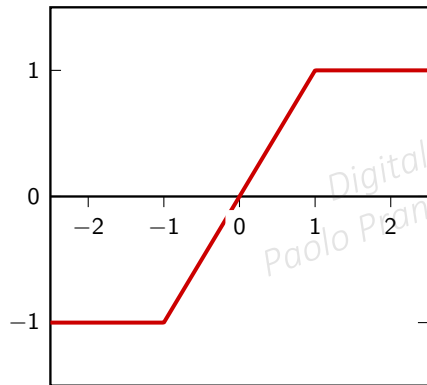
- ▶ clip samples to $[A, B]$: linear distortion (can be put to good use in guitar effects!)
- ▶ smoothly saturate input: this simulates the saturation curves of analog electronics

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If input is not uniform:

- ▶ use uniform quantizer and accept increased error.

For instance, if input is Gaussian:

$$\sigma_e^2 = \frac{\sqrt{3}\pi}{2} \sigma^2 \Delta^2$$

- ▶ design optimal quantizer for input distribution, if known (Lloyd-Max algorithm)
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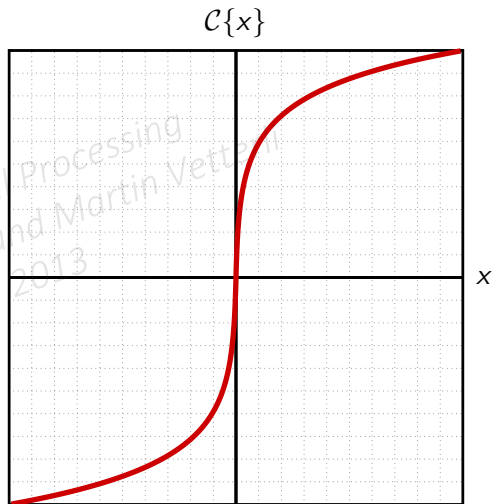
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$$\mathcal{C}\{x[n]\} = \text{sgn}(x[n]) \frac{\ln(1 + \mu|x[n]|)}{\ln(1 + \mu)}$$



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Module 7.3: A/D and D/A Conversion

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- ▶ Analog-to-digital (A/D) conversion

- ▶ Digital-to-analog (D/A) conversion

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- ▶ **sampling** discretizes time

- ▶ quantization discretized amplitude

- ▶ how is it done in practice?

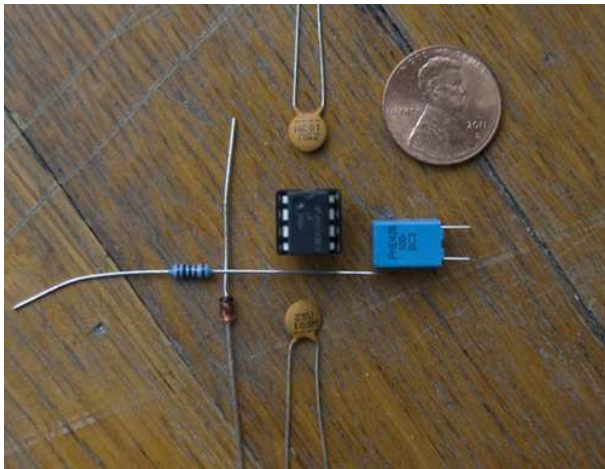
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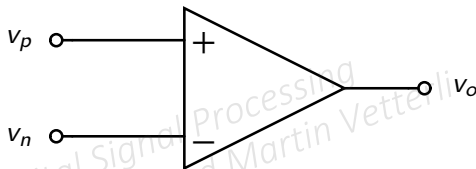
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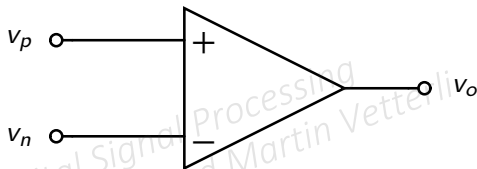
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$$v_o = G(v_p - v_n)$$



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The two key properties



- ▶ infinite input gain ($G \approx \infty$)
- ▶ zero input current

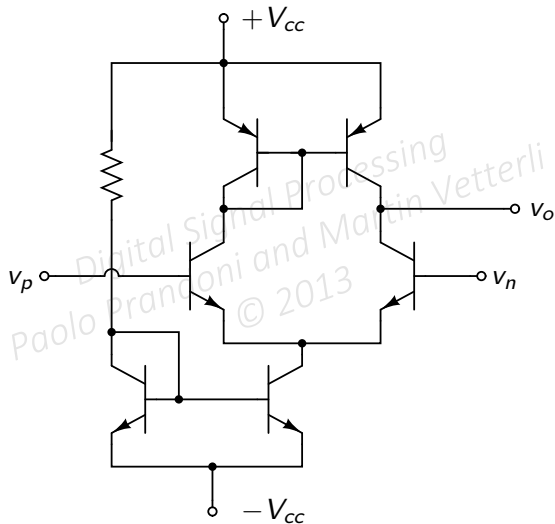
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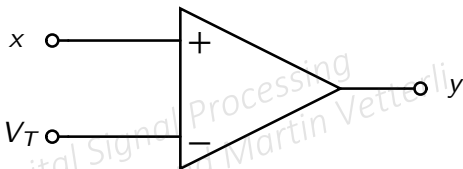


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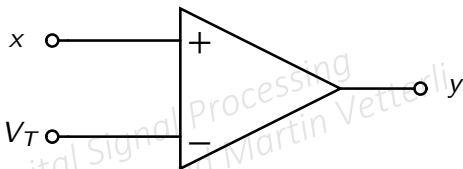


The op-amp in open loop: comparator



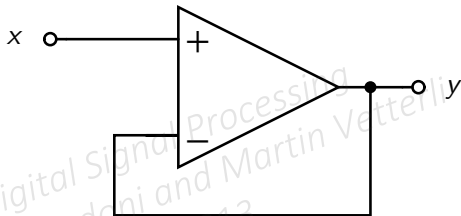
$$y = \begin{cases} +V_{cc} & \text{if } x > V_T \\ -V_{cc} & \text{if } x < V_T \end{cases}$$

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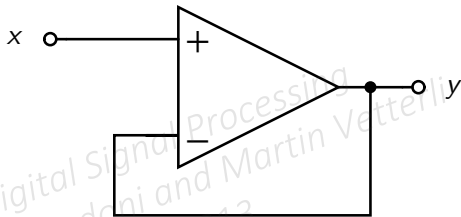
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The op-amp in closed loop: buffer



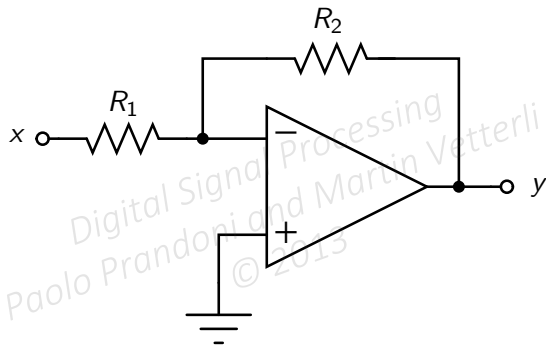
$$y = x$$

The op-amp in closed loop: buffer



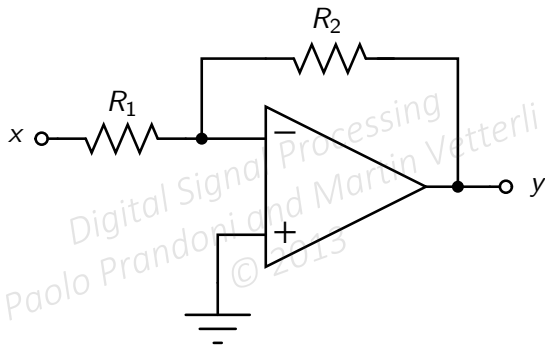
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The op-amp in closed loop: inverting amplifier

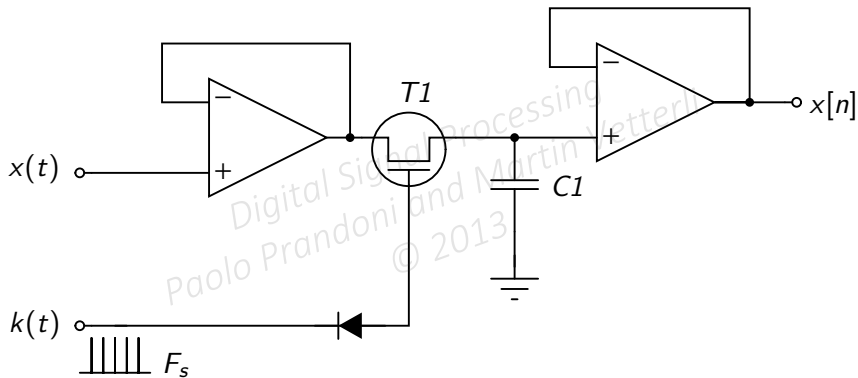


$$y = -(R_2/R_1)x$$

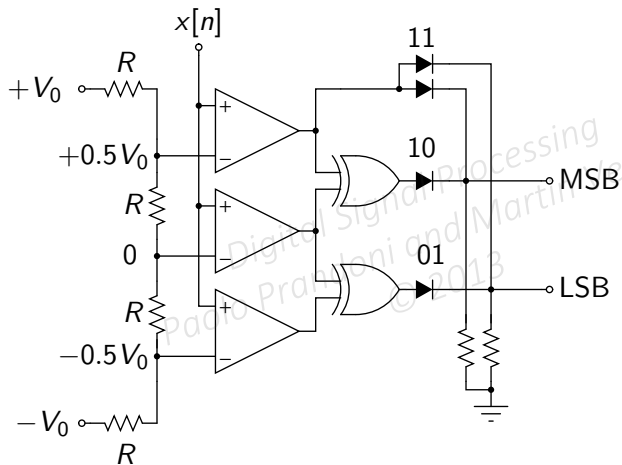
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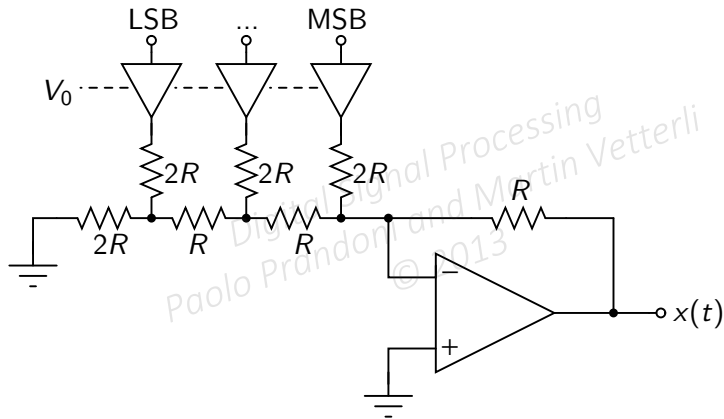


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A/D Converter: 2-Bit Quantizer





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