ESPTR
(English)
Signal Processing in Telecommunications and Radar

Jacek Misiurewicz
e-mail: jmisiure@elka.pw.edu.pl

Institute of Electronic Systems
Warsaw University of Technology
Warsaw, Poland

ESPTR: General information

"Credits" 2h/week lecture + 2h/week project.

Lecture Thursday, 08:15-10(Possible move to Tue, 08:15-10, probably room 162)
Team: dr J. Misiurewicz, dr K. Kulpa, mgr M. Malanowski

Contact J. Misiurewicz, (jmisiure@elka.pw.edu.pl) room 447. A web page is under construction (http://staff.elka.pw.edu.pl/˜jmisiure/esptr)

Project Simulation of a selected mechanism or technique. Three stages: definition, algorithm, program. Environment: Matlab or Octave, C/C++ (selected projects), other (special projects).

Tests One test within lecture hours (see the schedule).

Exam A final exam during the session

Scoring:

= 10%			test
		10%	P1: definition
	+	10%	P2: algorithm
	+	20%	P3: program
= 40%			Project total
= 50%			exam

Plan

- Some basics: frequency conversions, sampling&D/A, digital processing
- Radio channel, propagation, software radio, directional reception
- Radar basics, pulsed/CW radar, special radars
- Digital broadcasting and reception: DAB, DVB
- Cellular systems up to UMTS, structure, modulations, receivers

Basics: sampling

- ideal sampling
- non-ideal sampling: model as LP filter(conv)+sampling(mul), integrating AD converter case (multimeter)
- Nyquist sampling
- undersampling of narrowband signals (ideal and non-ideal case)
- reconstruction (ideal and nonideal)
- Oversampling to ease the antialiasing filter design

The Sampling Theorem

Named also after:

- 1915 Edmund T. Whittaker (UK)
- 1928 Harry Nyquist [ny:kvist] (SE)
- 1928 Karl Küpfmüller (DE)
- 1933 Vladimir A. Kotelnikov (USSR)
- 1946 Gábor Dénes (HU) $\longrightarrow$ Dennis Gabor (US)
- 1949 Claude E. Shannon (US)
- Cardinal Theorem of Interpolation Theory

Nyquist frequency, Nyquist rate

Sampling: bandlimited signal (aliasing problem)

Moiré pattern - as seen on TV, an exmaple of too low sampling frequency.

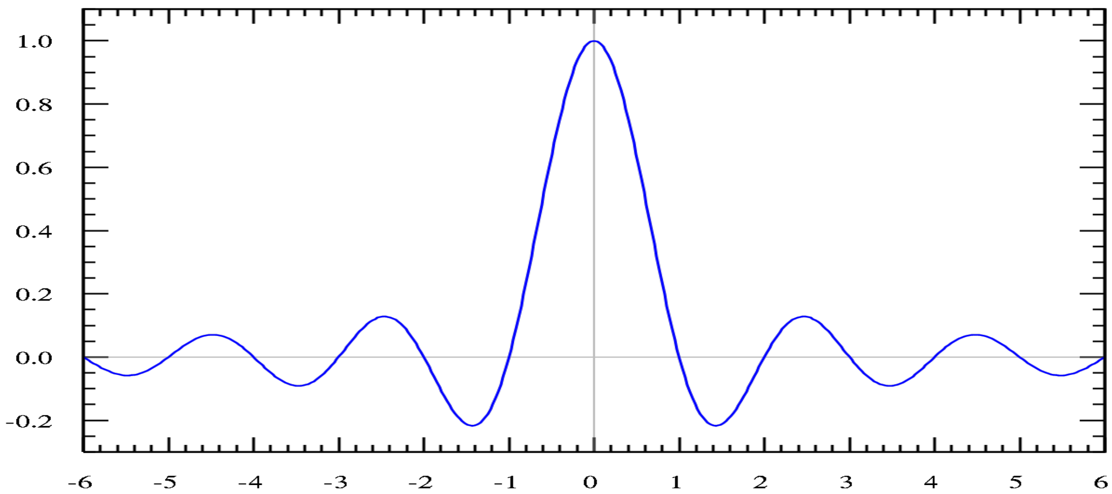
Reconstruction

Reconstruction: interpolation,

$$x(t) = \sum_{n=-\infty}^{\infty} x[n] \cdot \operatorname{sinc}\left(\frac{t - nT}{T}\right)$$

lowpass filtering

$$x(t) = \left(\sum_{n=-\infty}^{\infty} x[n] \cdot \delta(t - nT)\right) * \operatorname{sinc}\left(\frac{t}{T}\right)$$



Bandpass sampling

Bandwidth less than $f_s \longrightarrow$ e.g. a signal in the band $Nf_s \pm 0.5f_s$

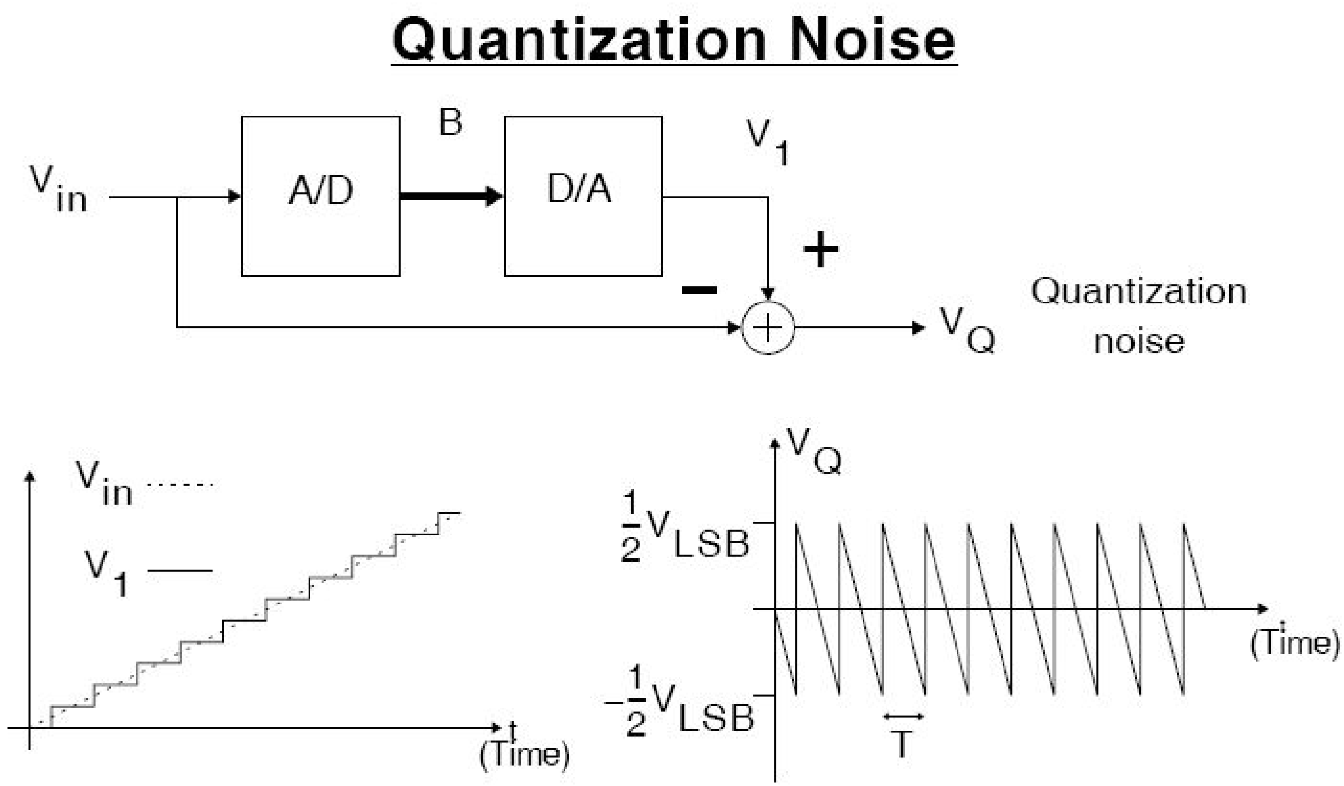
Antialiasing filter: bandpass!

Reconstruction: with bandpass filter!

The influence of non-ideal sampling (*system bandwidth*) $\longrightarrow$ unwanted lowpass filter.

Sampling jitter problem: present with Nyquist sampling, much sharper with bandpass sampling.

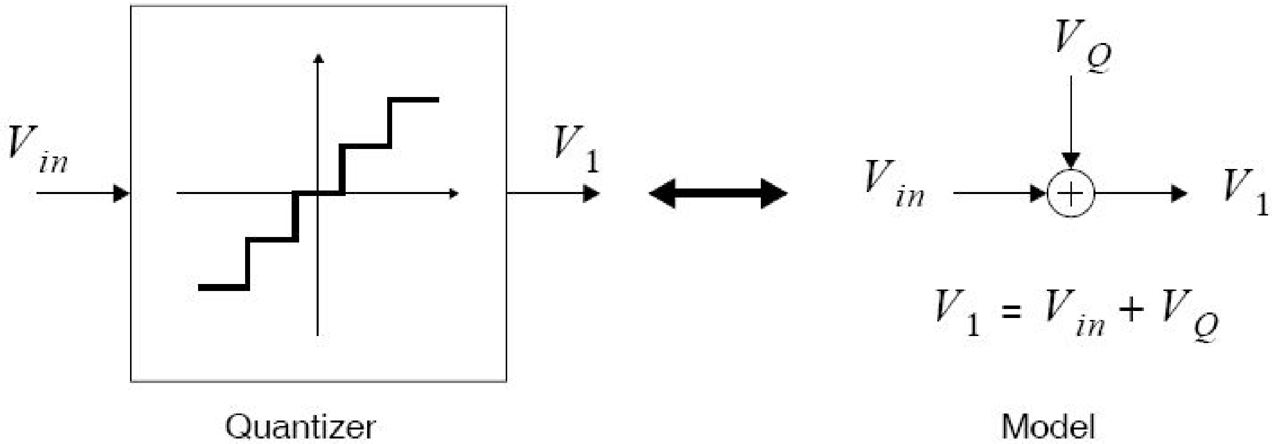
A/D noise



$$V_Q = V_1 - V_{in} \quad \text{or} \quad V_1 = V_{in} + V_Q \tag{8}$$

A/D noise

Quantization Noise



- Above model is exact
  — approx made when assumptions made about V_Q
- Often assume V_Q is white, uniformily distributed number between $\pm V_{\text{LSB}}/2$

A/D SNR

Noise amplitude: $q/2$, assumed uniformly distributed $\longrightarrow \sigma_n^2 = \frac{q^2}{12}$ (power).

Each extra bit gives 2x smaller $q \longrightarrow$ 6.02 dB less noise.

SNR with assumption that “signal” is a maximum-amplitude $(2^N \cdot q)$ sinusoid:

$$SNR = 10\log_{10}\frac{\text{signal power}}{\text{noise power}}[dB] = 10\log_{10}\frac{(2^N q)^2/(2{\cdot}2^2)}{q^2/12}[dB] =$$

$$= 10\log_{10}(1.5 * 4^N)[dB] = 1.76 + 6.02 \cdot N[dB]$$

Oversampling

- More space for transition band of antialiasing filter (A/D) or reconstruction filter (D/A)
- (AD) later we may LP filter and downsample signal: we gain 1 bit of accuracy for a 4-sample average; more gain with *noise shaping* $\longrightarrow$ sigma-delta converters (not discussed further at ESPTR)

D/A

- speed
- bits
- nonidealities