ESPTR
(English)
Signal Processing in Telecommunications and Radar

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ESPTR: General information

“Credits” 2h/week lecture + 2h/week project.

Lecture Thursday, 08:15-10(Possible move to Tue, 08:15-10, probably room 122)
Team: dr J. Misiurewicz, mgr A. Gromek, mgr P. Krysik

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

Projects Simulation of a selected mechanism or technique. **Two** projects (“R” - Radio/Radar, due mid-term, “T” - Telecomm, due before end of term), each with two stages: 1. definition, 2. final. Environment: Matlab, Octave or NumPy, C/C++ (selected projects), other (special projects).

Exam A final exam during the session

Scoring:

	5%	proj. R1: definition	
+	20%	proj. R2: final	**Minimum 25 pt from projects is**
+	5%	proj. T1: definition	**necessary to get score** > 3
+	20%	proj. T2: final	See webpage for project deadlines
=	50%	Project total	and penalties for late submission.
=	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 4G/LTE, 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) $\longrightarrow$ (US)
- 1928 Karl Küpfmüller (DE)
- 1933 Vladimir A. Kotelnikov (USSR)
- 1946 Gábor Dénes (HU) $\longrightarrow$ Dennis Gabor (UK)
- 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.

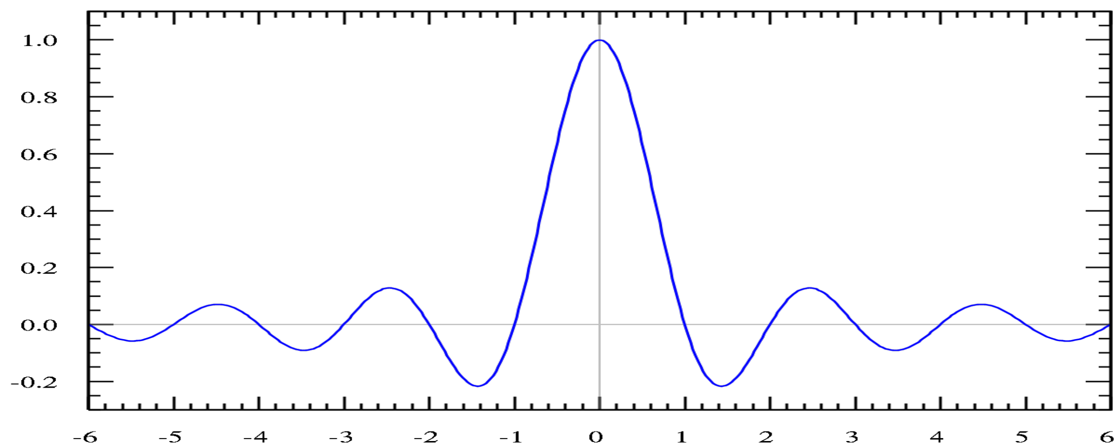
Reconstruction

Reconstruction: interpolation, (*sinus cardinalis* $\mathrm{sinc} = \mathrm{Sa} = \frac{\sin(\pi x)}{\pi x} = \mathrm{j}_0(\pi x)$)

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

lowpass filtering (Küpfmüller filter) (DE)

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



Bandpass sampling

(sometimes called “undersampling”)

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

Antialiasing filter: bandpass!

Reconstruction: with bandpass filter!

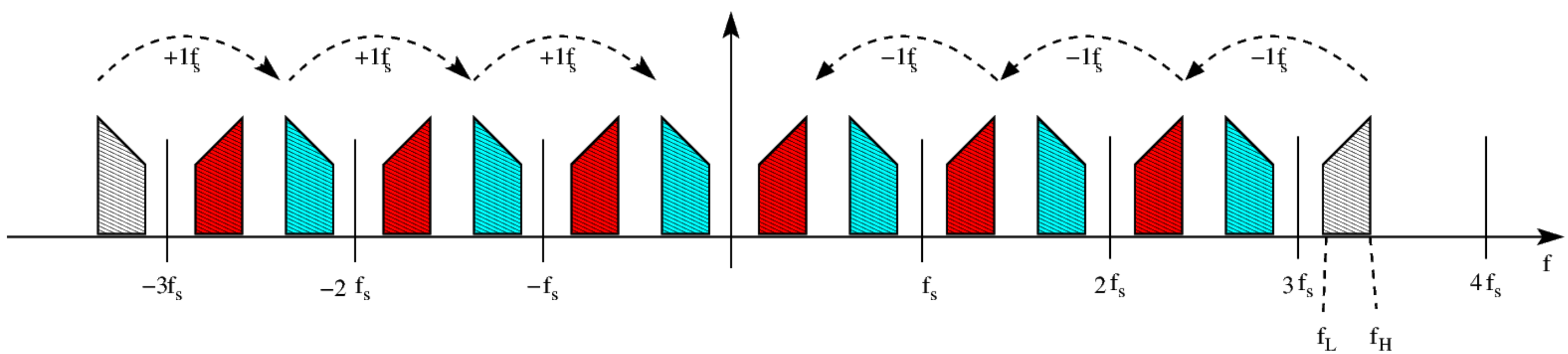
The influence of non-ideal sampling (*system bandwidth*) $\longrightarrow$ unwanted lowpass filter.

Sampling jitter problem: present with Nyquist sampling, much sharper with bandpass sampling. (steeper slope of the signal at the sampling point...)

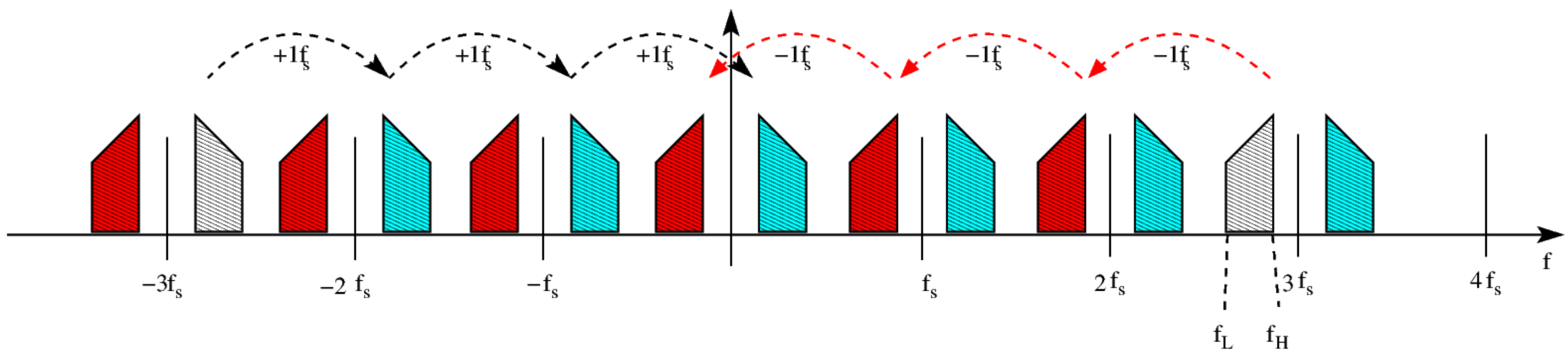
Bandpass sampling - frequency choice

Signal frequency from f_L to f_H. (Assume real-valued signals, so actually from $-f_H$ to $-f_L$ and from f_L to f_H).
Bandwidth $B = f_H - f_L$ (one-sided...)

- Baseband sampling: $f_L \geq 0$, $f_H < f_s/2$
- N times undersampling $N \leq \frac{f_H}{B}$
- Avoid aliasing: $f_s \frac{N}{2} \geq f_H$, $f_s \frac{N-1}{2} \leq f_L$
- N even - spectrum mirroring!

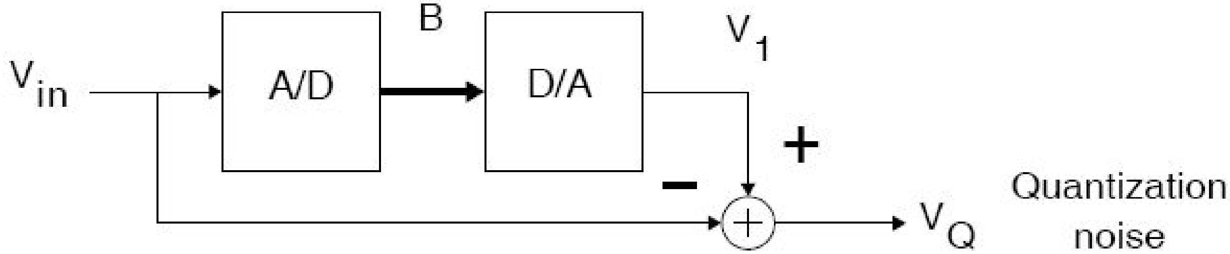


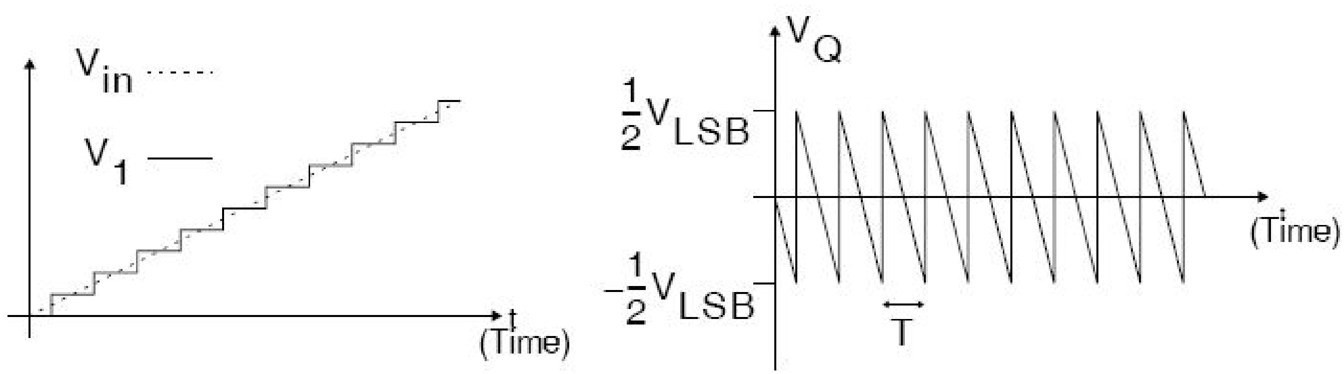
Bandpass sampling - mirroring



A/D noise

Quantization Noise

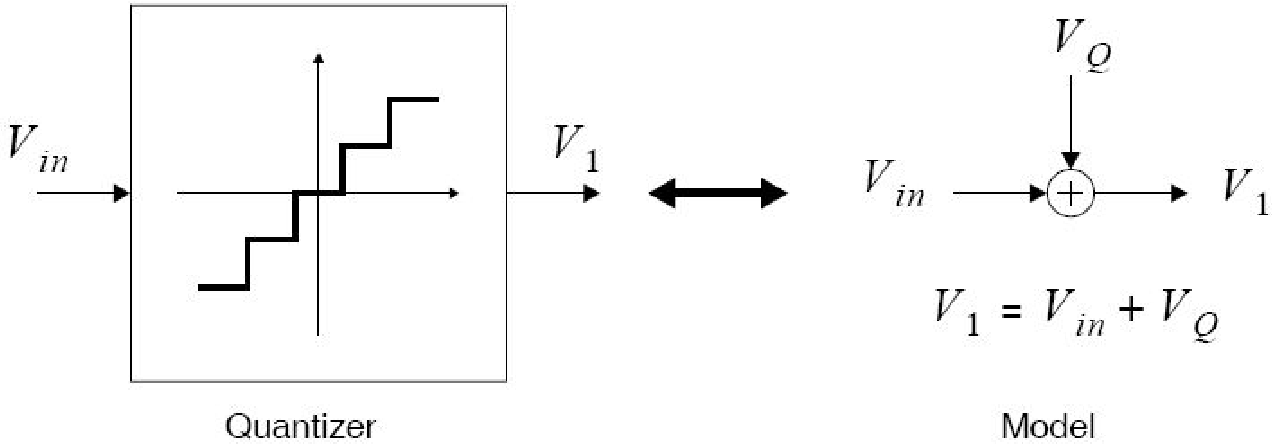




$$V_Q = V_1 - V_{in} \quad \text{or} \quad V_1 = V_{in} + V_Q \tag{8}$$

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 $\pm V_{\mathrm{LSB}}/2$

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 ($V_{pp} = 2^N \cdot q$, and $power(sinusoid) = (V_{pp}/2)^2/2$) 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)

A/D types

- Multimeters: integrating (e.g. dual slope) = voltage to time conversion + time measurement
- Slow signals: ramp voltage or successive approximation (D/A + comparator + control circuit)
- Audio: usually sigma-delta (oversampling with 1/bit conversion and noise shaping + LP filter + decimation)
- Video: flash (a ref voltage ladder + a lot of comparators)
- Combo: subranging convereter = more stages (rough AD + DA + next AD on the remainder)

D/A and A/D problems

- speed
- bits (from 6 to 24) - remember that this may be noise-limited or q limited...
- nonidealities