Lecture 4
Biosignal Processing

Digital Signal Processing and Analysis in Biomedical Systems

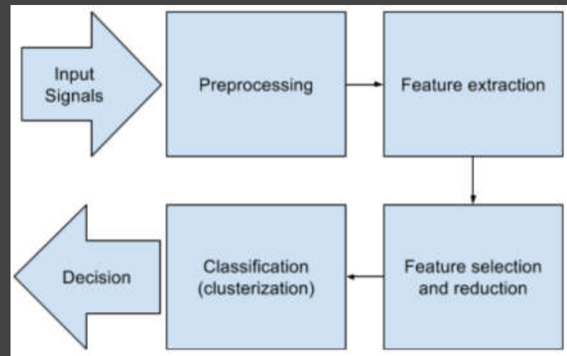
Contents

- Preprocessing as first step of signal analysis
- Biosignal acquisition
- ADC
- Filtration (linear, Wiener, Kalman, Savitzky–Golay, adaptive)

Useful parameters of biosignals

1. Waveform shape
2. Rate of change (spectral content)
3. Magnitude/duration
4. Onset/offset of events
5. Similarity/synchronicity with other signals

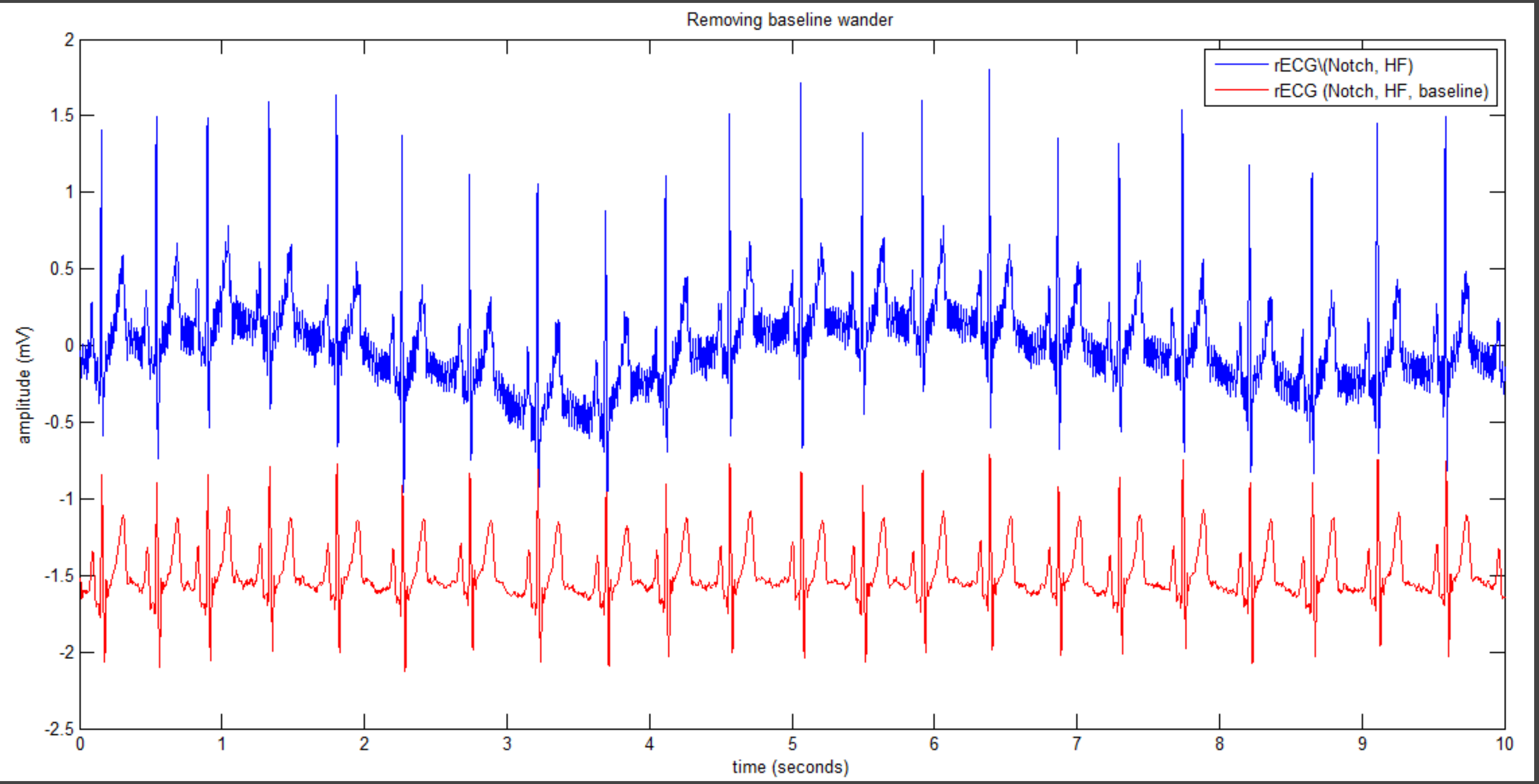
Biomedical signal analysis steps



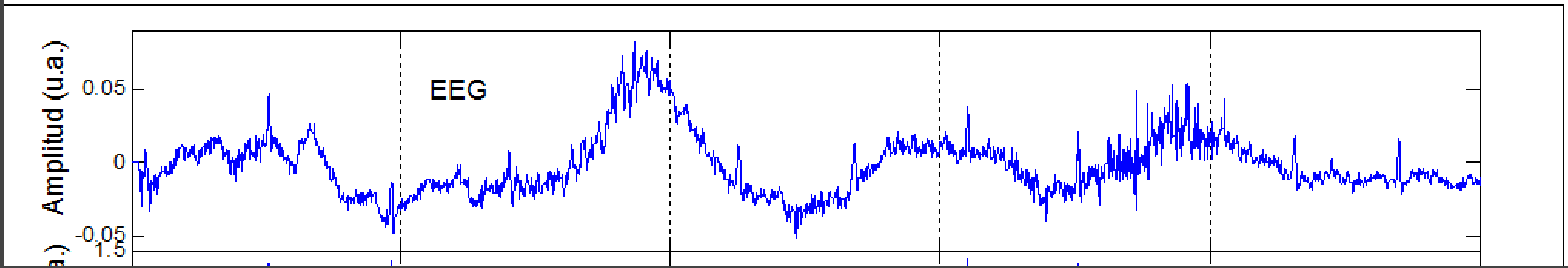
Physiological information often cannot be available directly from the raw recorded signals:
- it can be masked by other biologic signals (artefacts),
- or hidden in noise.

Additional (preliminary) processing is usually required to enhance the relevant information and to extract from it the parameters that quantify the behavior of the system under study.

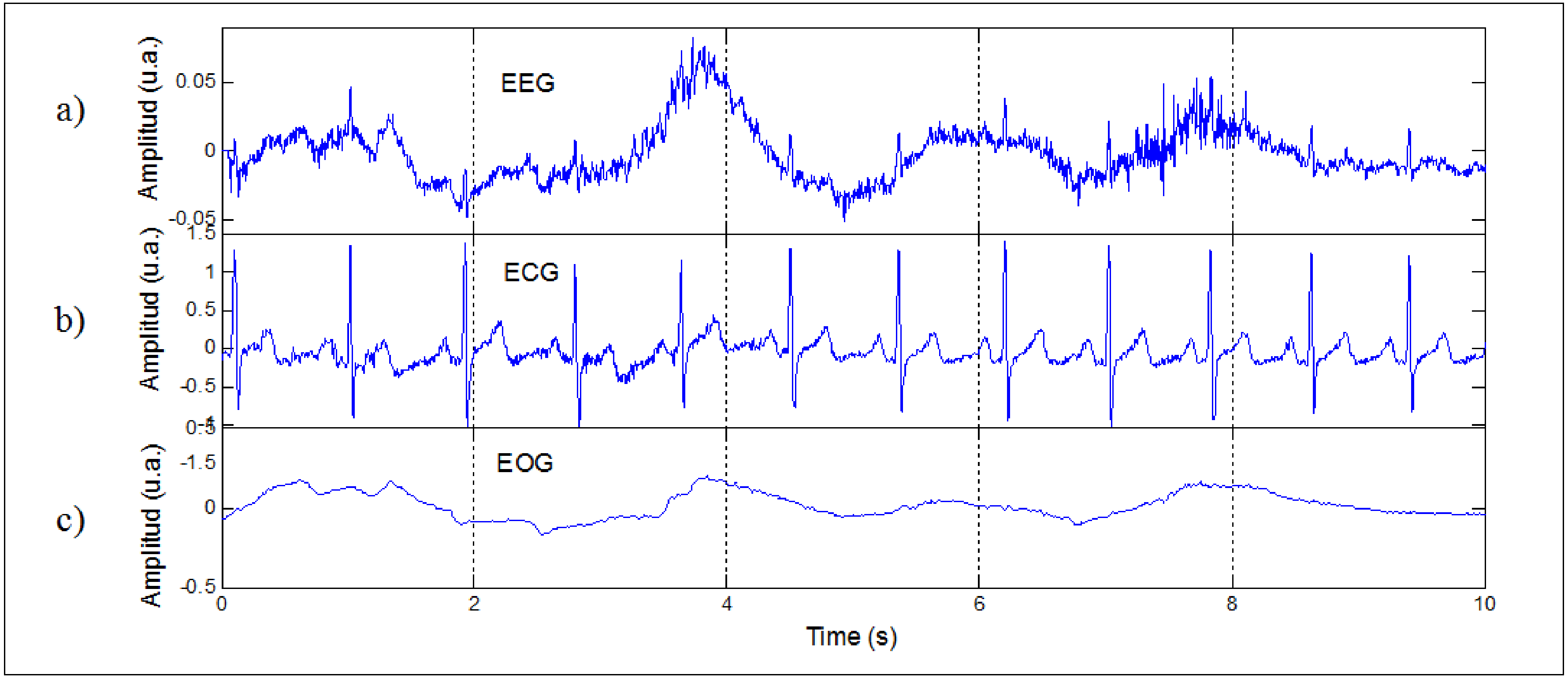
Noisy ECG


Removing baseline wander
rECG\(Notch, HF)
rECG (Notch, HF, baseline)
amplitude (mV)
time (seconds)
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
-2.5
0
1
2
3
4
5
6
7
8
9
10

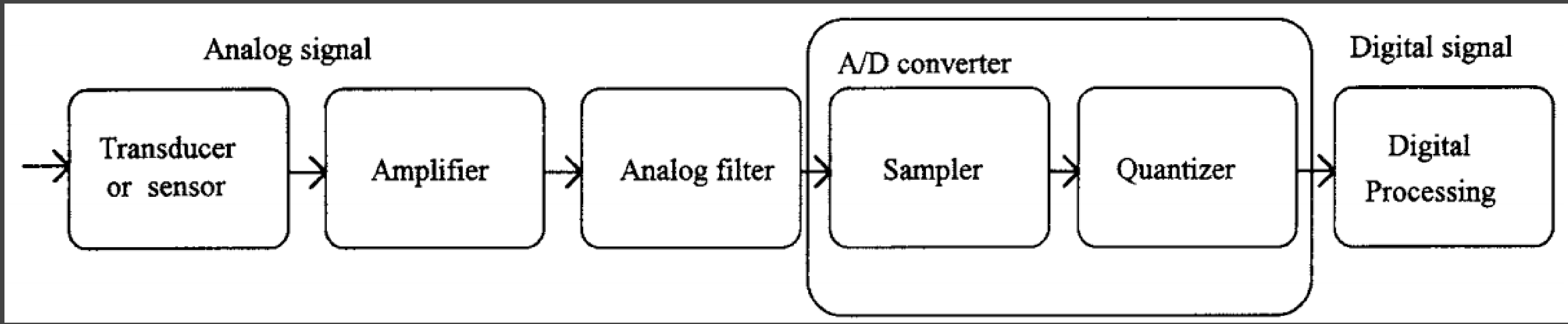
EEG with low-frequency component and spikes



EEG corrupted with ECG and EOG



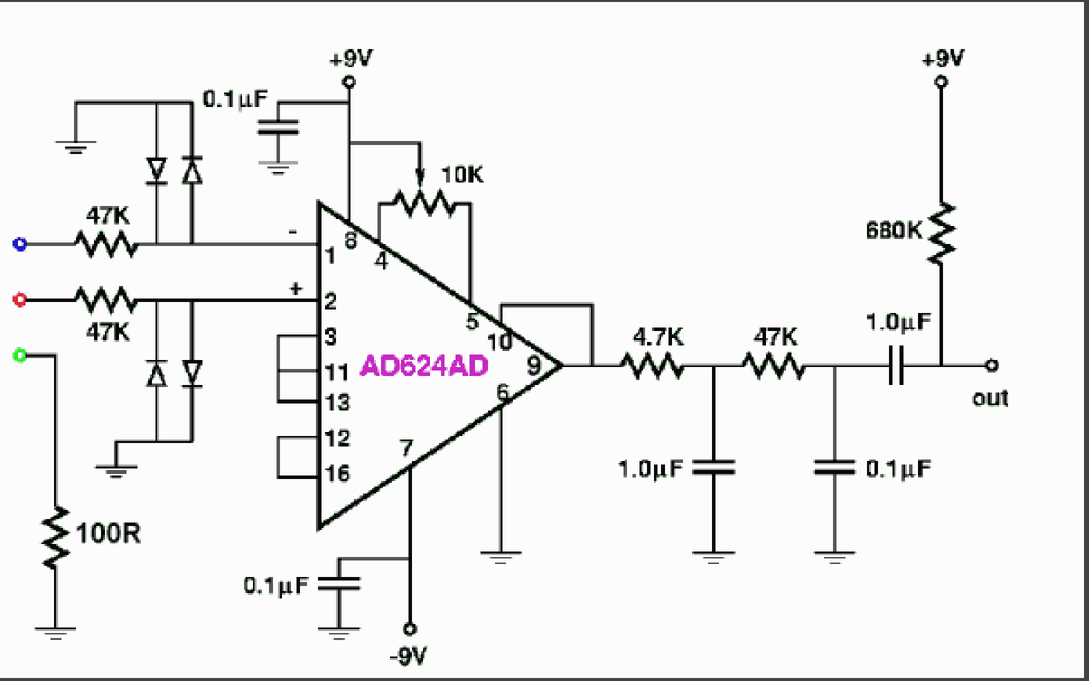
Biosignal processing block diagram



Processing – changing the characteristics of a signal.

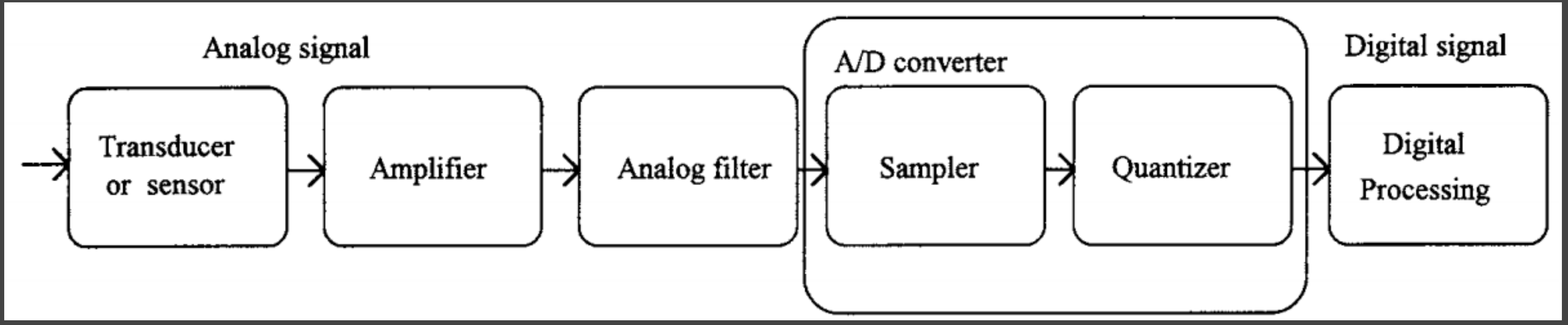
Analog signal processing

- Detection (protection)
- Amplification
- Filtration (noise suppression, baseline wandering removal)
- Change of the baseline
- Change of dynamic range



Digital signal processing

- Sampling
- *Segmentation*
- *Accumulation (ECG late potential detection)*
- *Averaging (Evoked response)*
- Digital Filtration

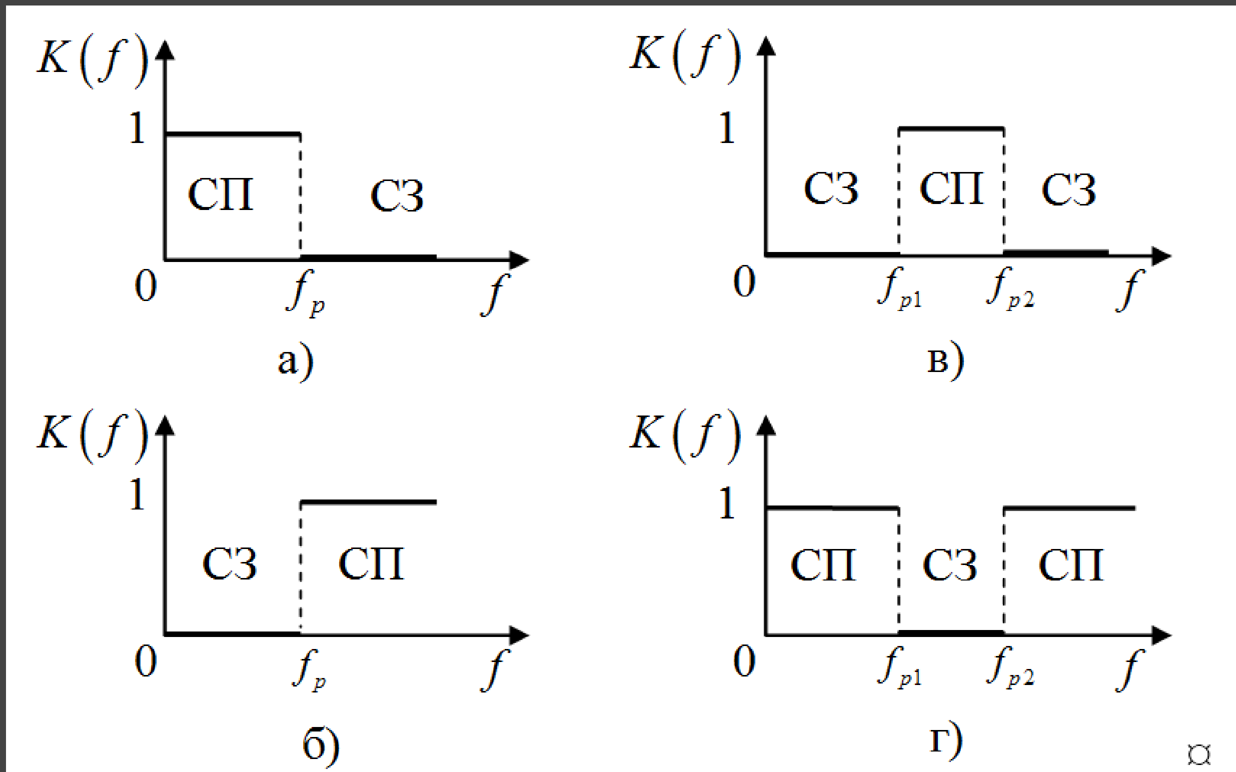


Filtration

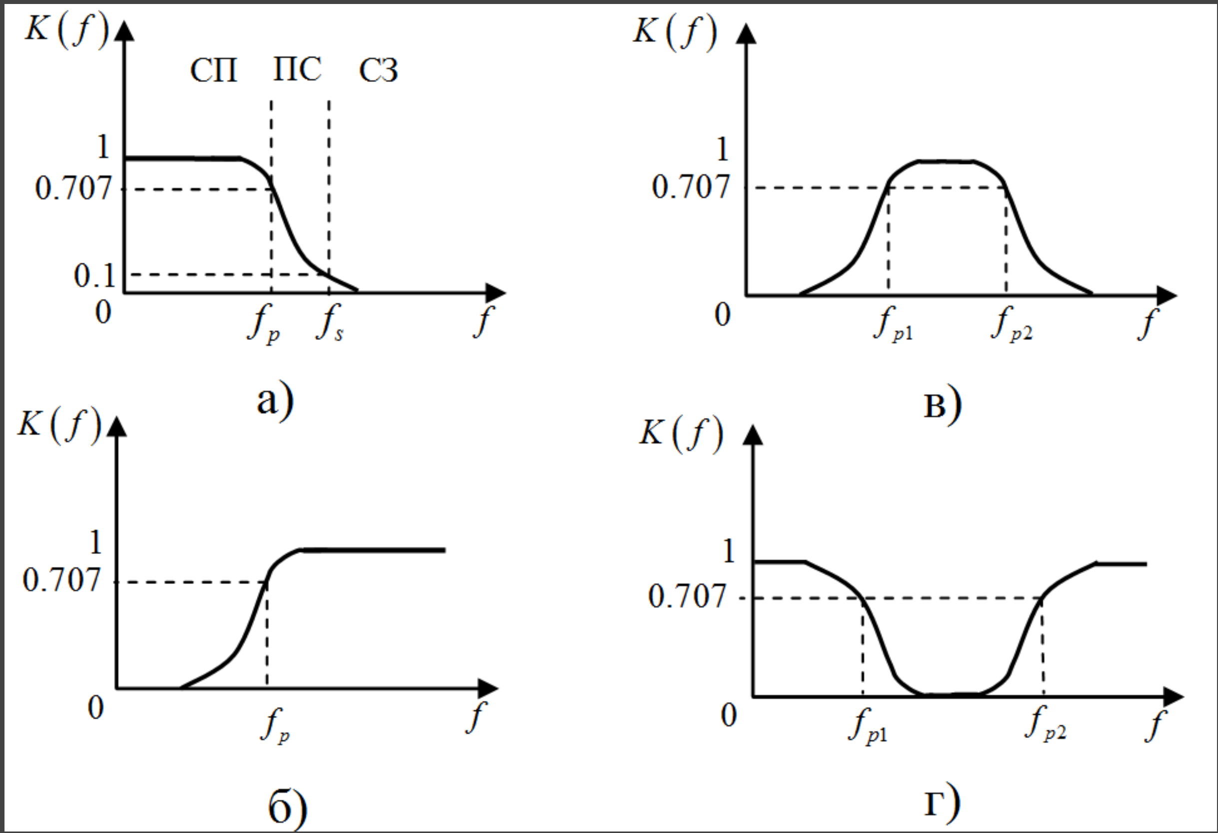
Filter is a system that suppresses or removes some unwanted component or feature from a signal. Most often, filtration – the change of spectral content of a signal. Filters:

- linear or non-linear,
- time-invariant or time-variant,
- causal or not-causal,
- analog or digital,
- discrete-time (sampled) or continuous-time,
- passive or active type of continuous-time filter
- infinite impulse response (IIR) or finite impulse response (FIR) type of discrete-time or digital filter.

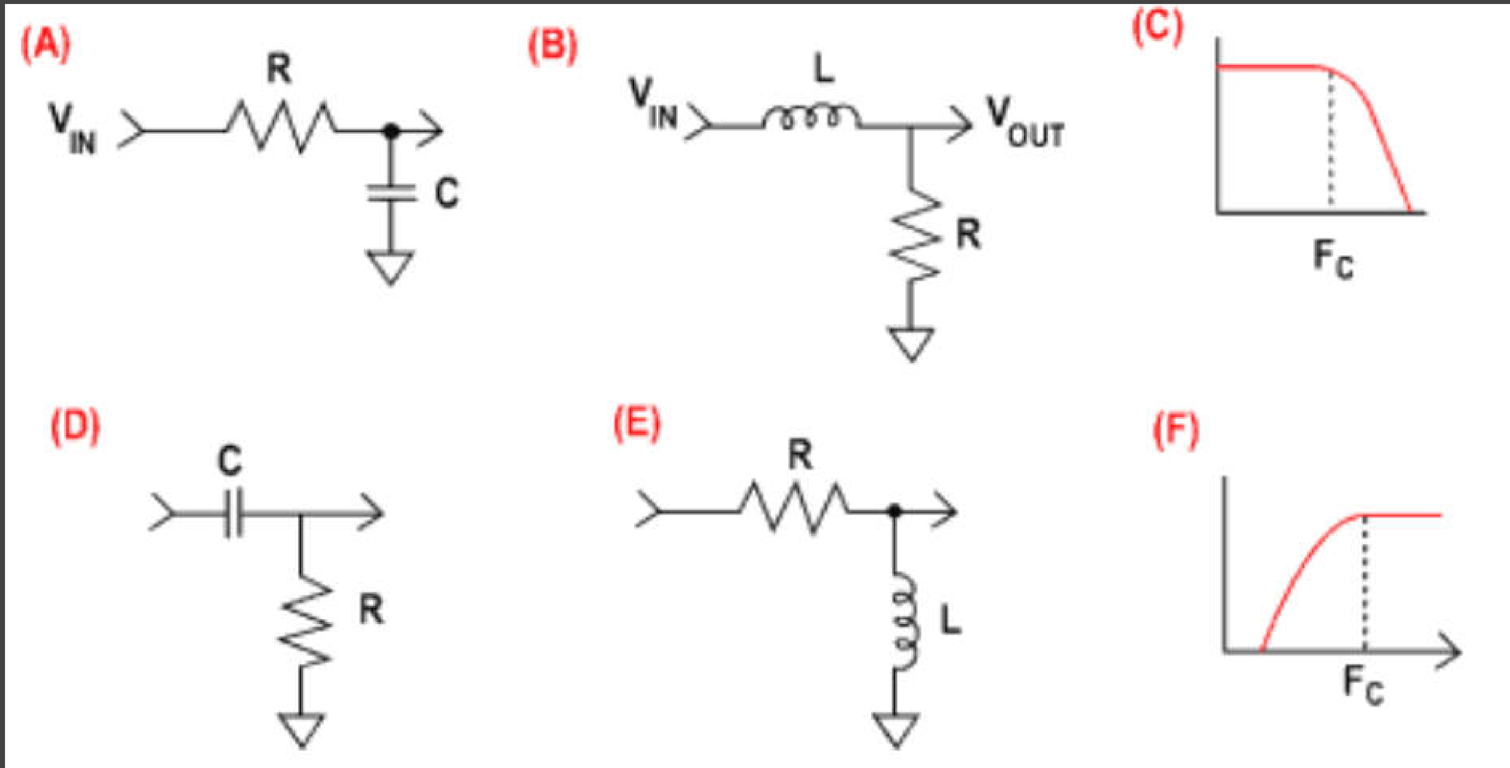
Analog (continuous) filters (ideal)



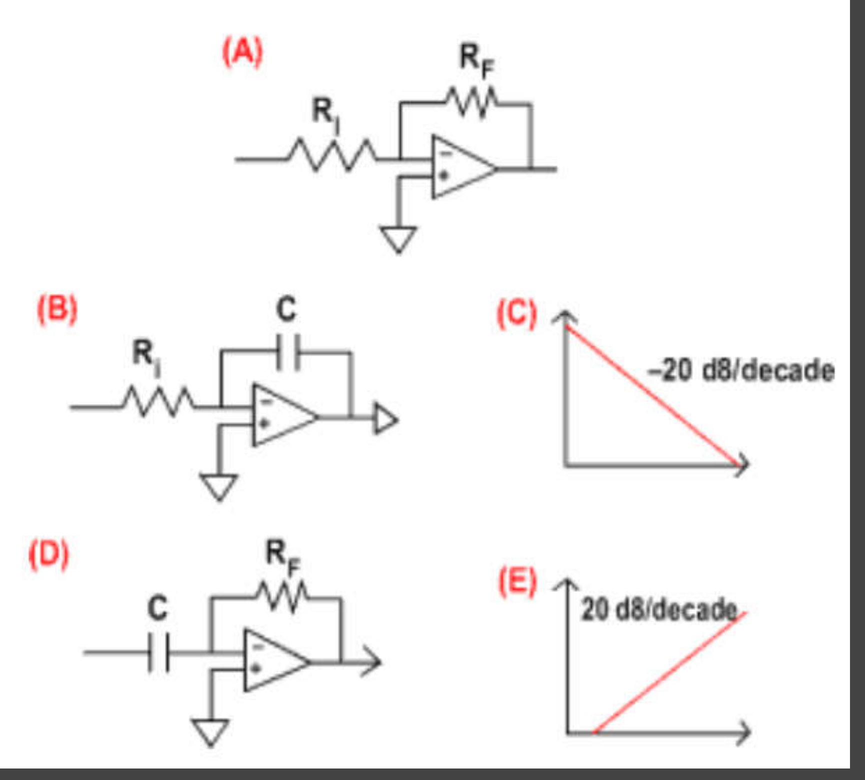
Analog (continuous) filters (real)



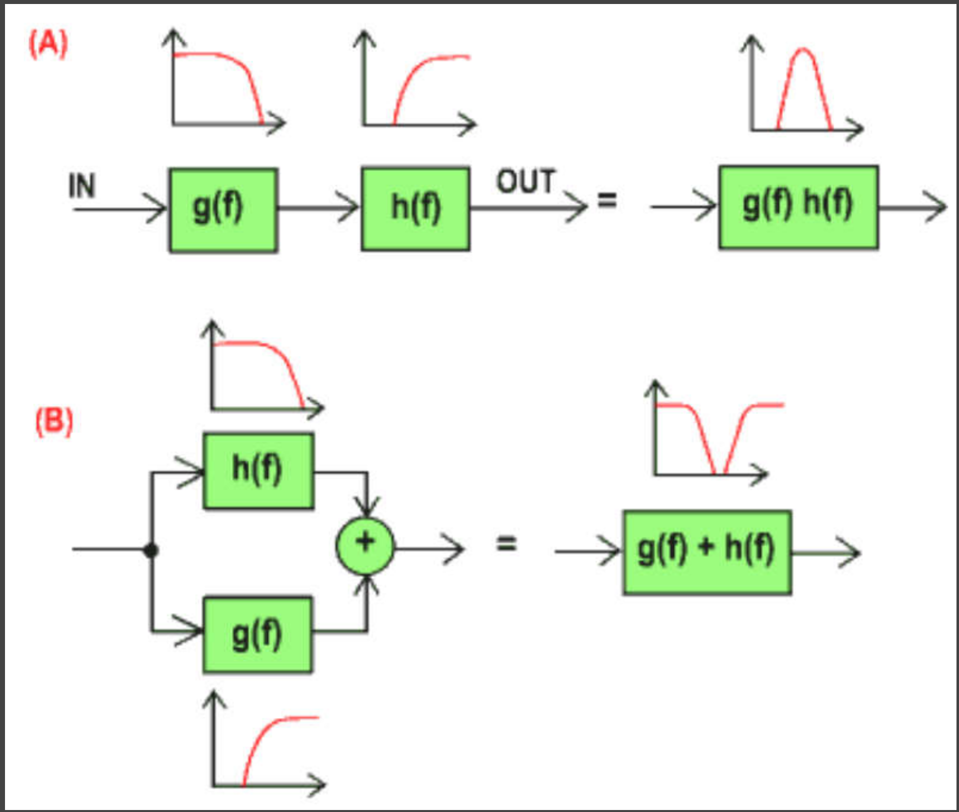
Basic schemes of passive analog filters



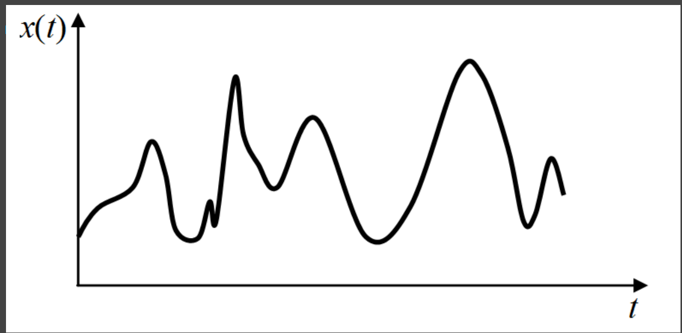
Basic schemes of active analog filters

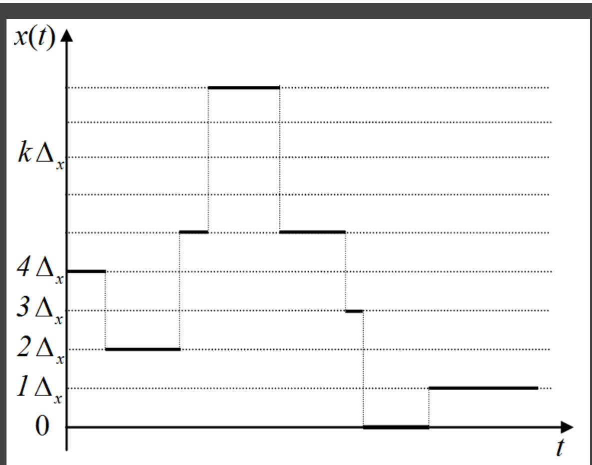


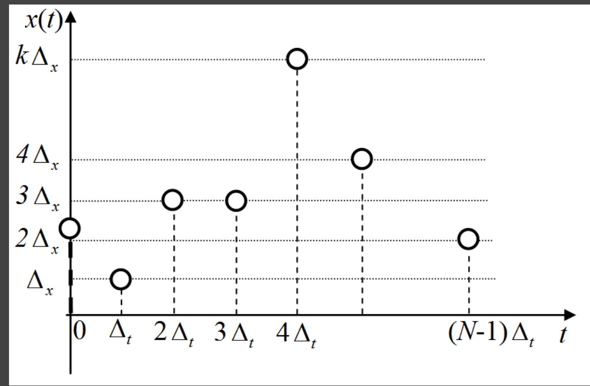
Combining simple filters

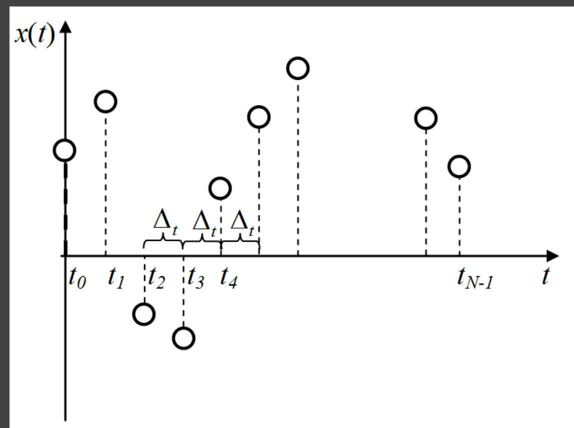


Four types of signals



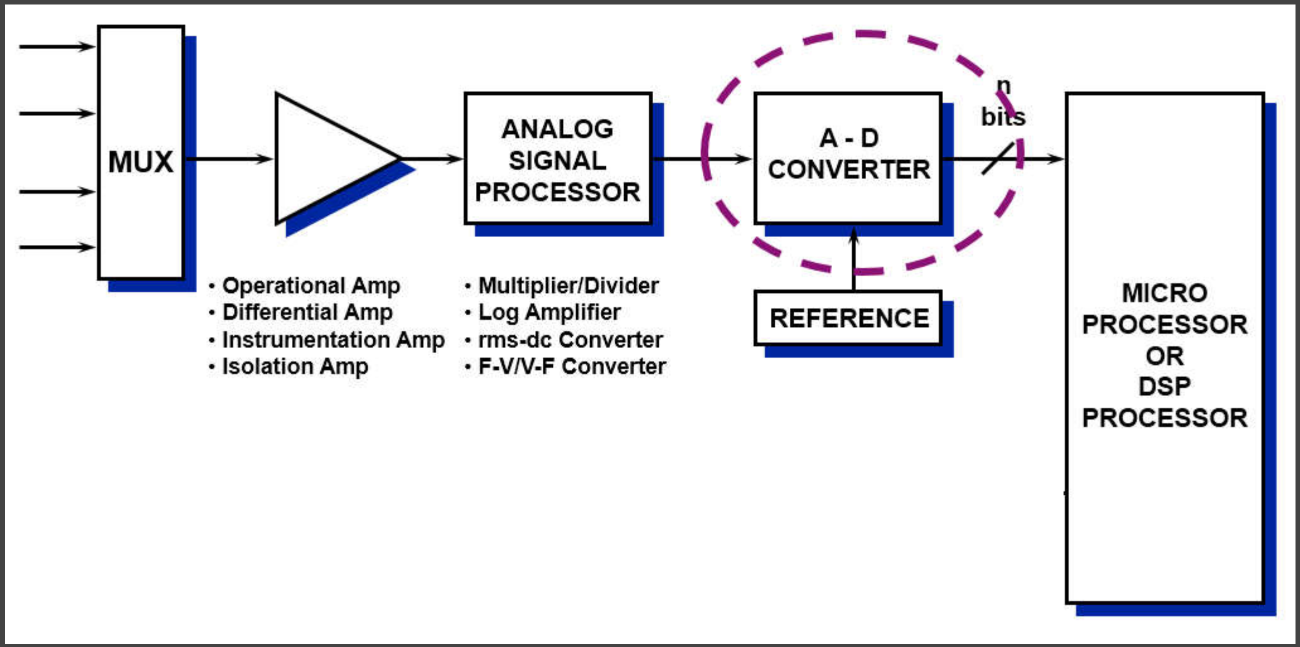




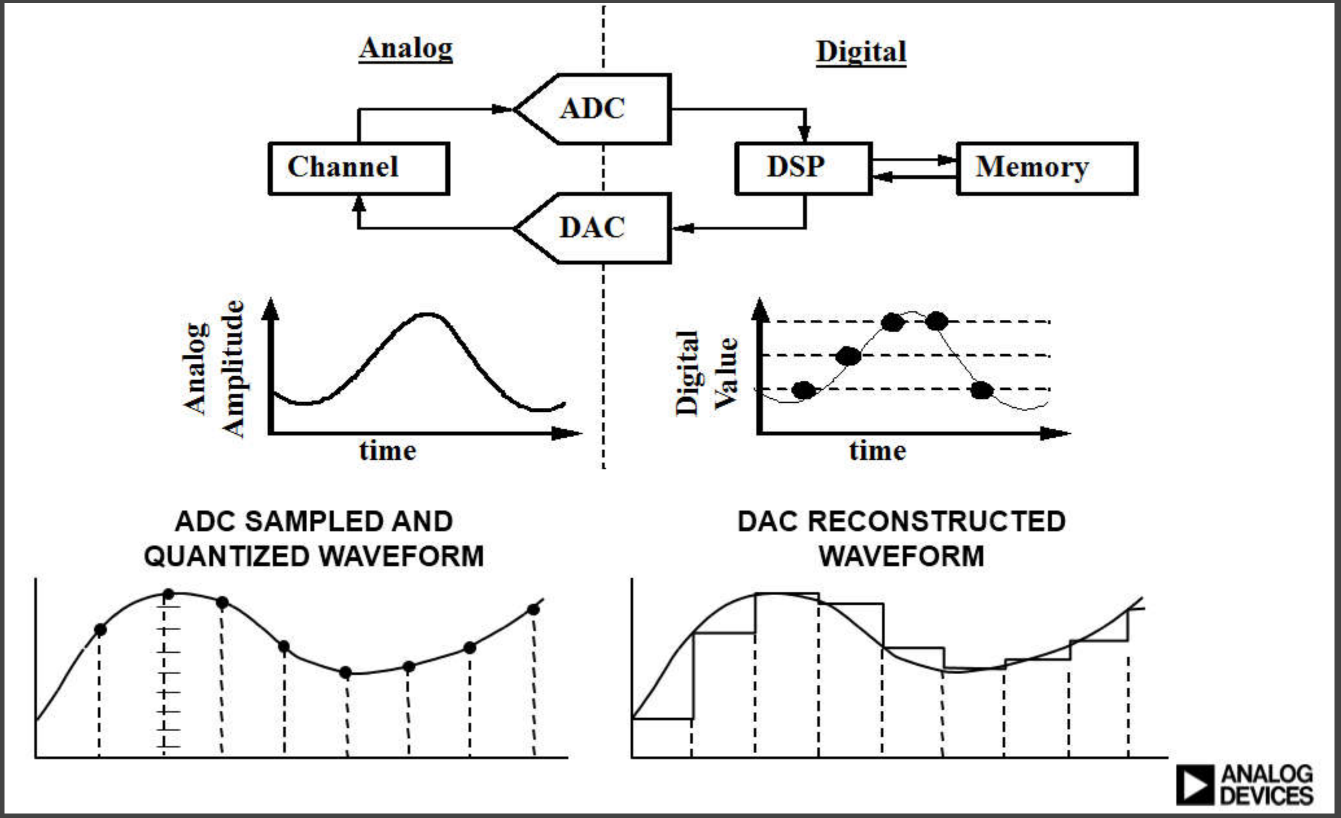


Analog-to-digital conversion

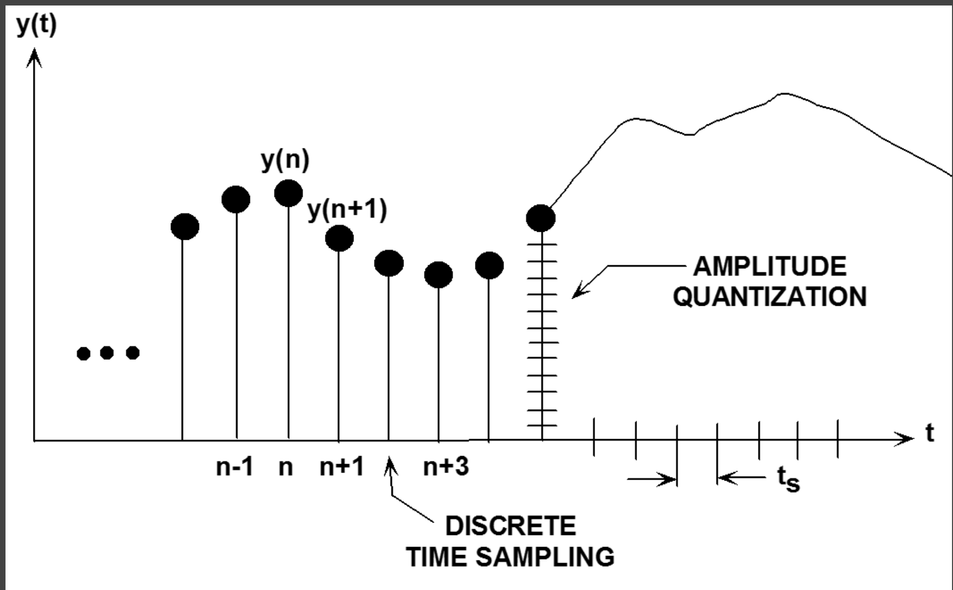
ADC converts analog signals into binary code.

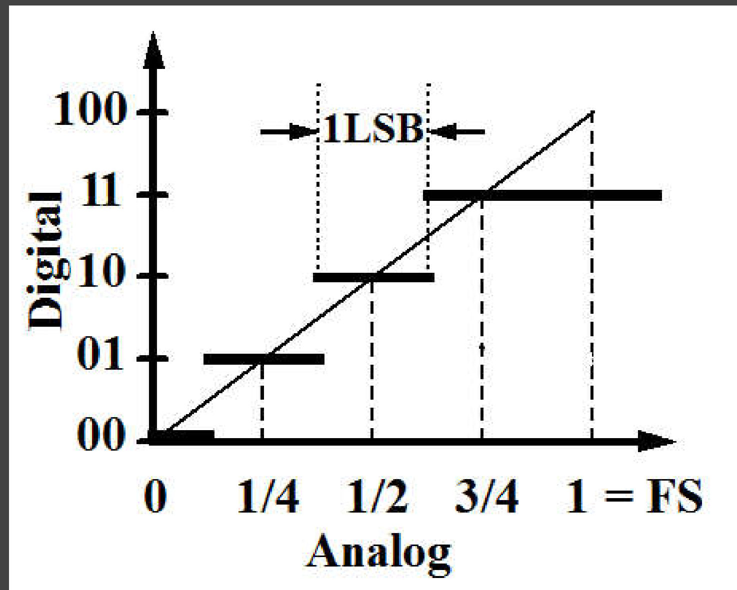


Real digital system

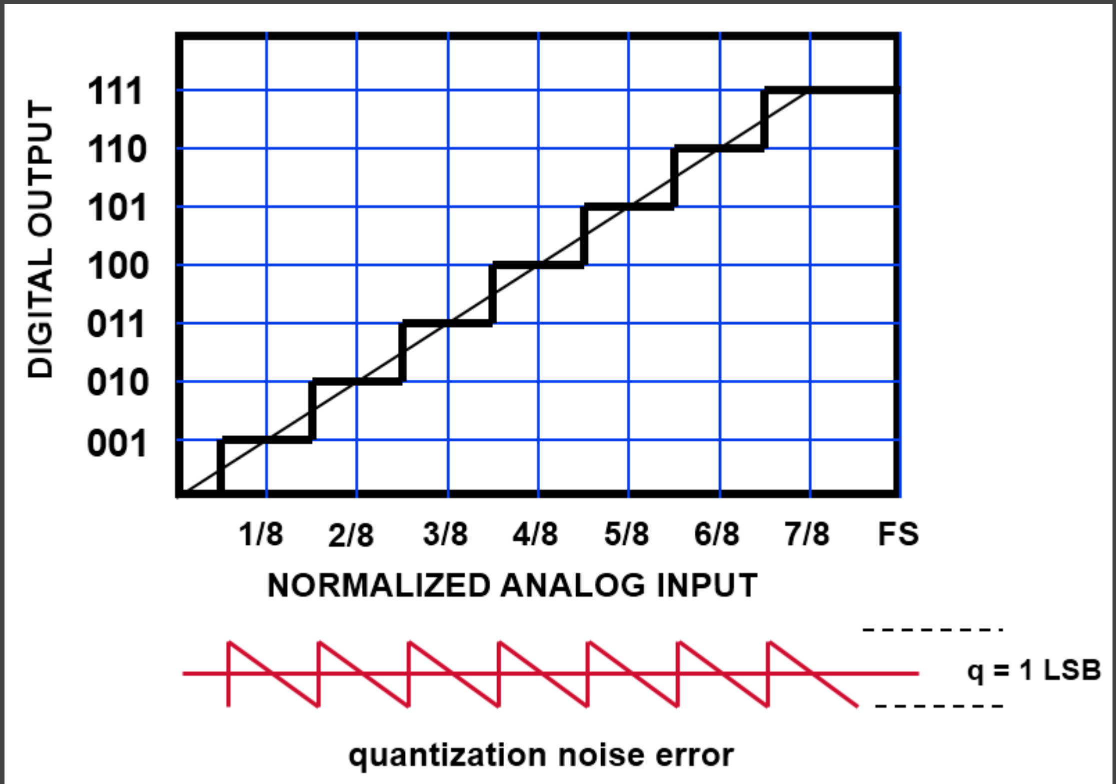


Sampling and Quantization





Quantization noise



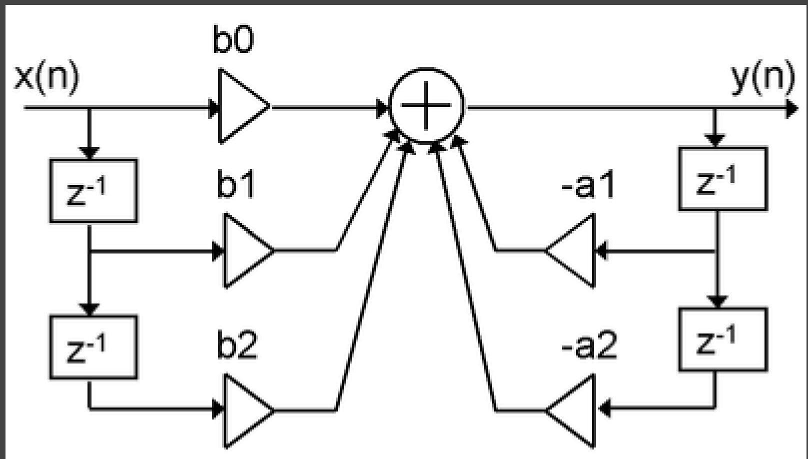
Digital filters

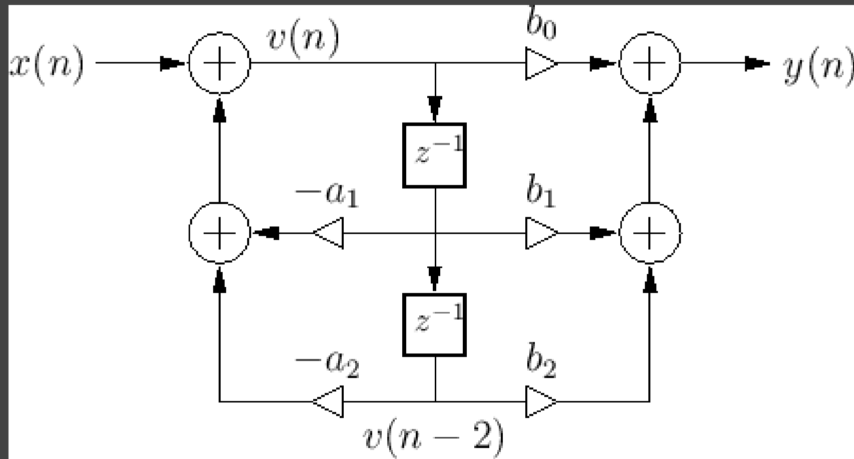
Digital filter is a system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal.

$$\sum_{k=0}^{N-1} a_k y[n-k] = \sum_{m=0}^{M-1} b_m x[n-m],$$

$$y[n] = \sum_{m=0}^{M-1} b_m x[n-m] - \sum_{k=1}^{N-1} a_k y[n-k].$$

Filter structures

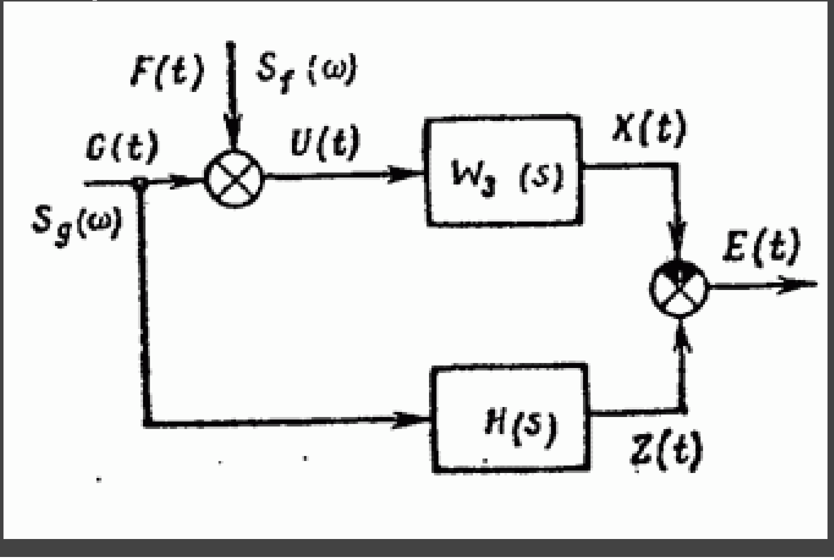




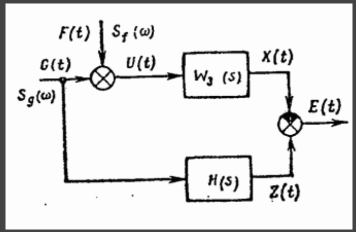
Optimal filtration – Wiener filter

Wiener filter is a filter used to produce an estimate of a random signal by linear time-invariant (LTI) filtering of an observed noisy process, assuming known stationary signal and additive noise spectra.

The Wiener filter designing procedure minimizes the mean square error between the estimated random process and the desired process.



Wiener–Hopf equation



Impulse response:

$$\int_{-\infty}^{+\infty} h_{opt}(t) R_u(\tau - t) dt - R_{zu}(\tau) = 0, \qquad \tau \geq 0 .$$

Applicable when signal and noise are correlated zero-mean stationary processes.

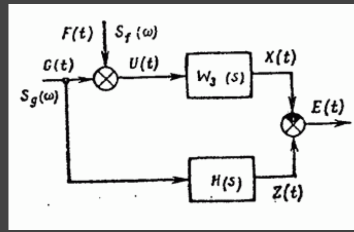
$$R_u(\tau) = R_g(\tau) + R_f(\tau) + R_{gf}(\tau) + R_{fg}(\tau)$$

Correlation of the input signal

$$R_{zu}(\tau) = \lim_{T \to \infty} \frac{1}{2T} \int_{-T}^{T} z(t) u(t + \tau) dt$$

Correlation of the input and output signals

Wiener optimal filter



$$W_{opt}(j\omega) = \frac{1}{2\pi\Psi(j\omega)} \int_{0}^{\infty} e^{-j\omega t} dt \int_{-\infty}^{\infty} \frac{S_{zu}(j\omega)}{\Psi(-j\omega)} e^{j\omega t} d\omega$$

Transfer function in frequency domain

$$S_{zu}(j\omega) = S_{zg}(j\omega) + S_{zf}(j\omega)$$

Spectral density of input and output signals

$$\Psi(j\omega)\Psi(-j\omega) = \left|\Psi(j\omega)^2\right| = S_u(\omega) = S_g(\omega) + S_f(\omega) + S_{gf}(\omega) + S_{fg}(\omega)$$

$$S_{gf}(j\omega) = S_{fg}(j\omega) = S_{zf}(j\omega) = 0$$

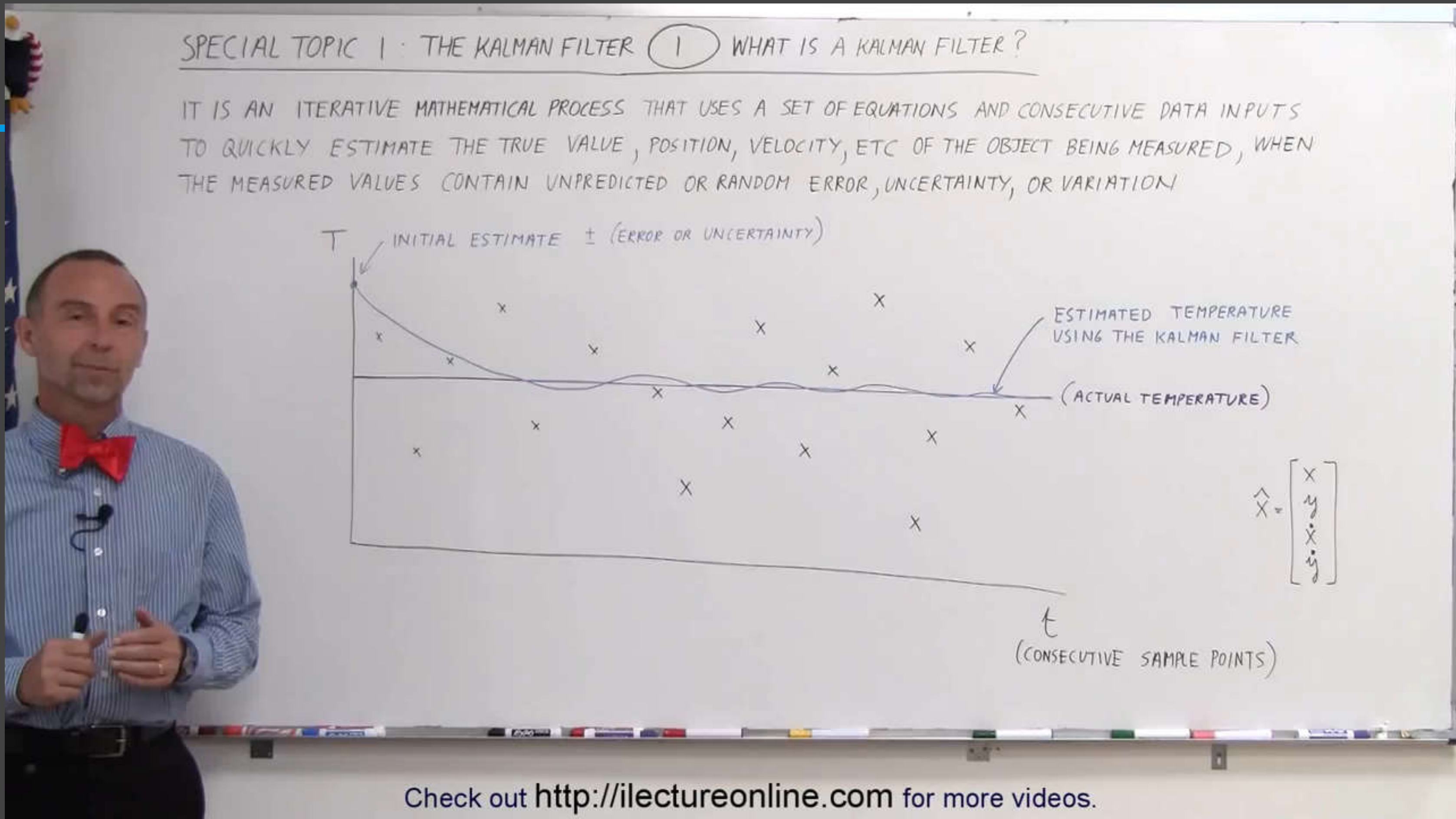
Applicable when signal and noise are non-correlated!

Kalman filter

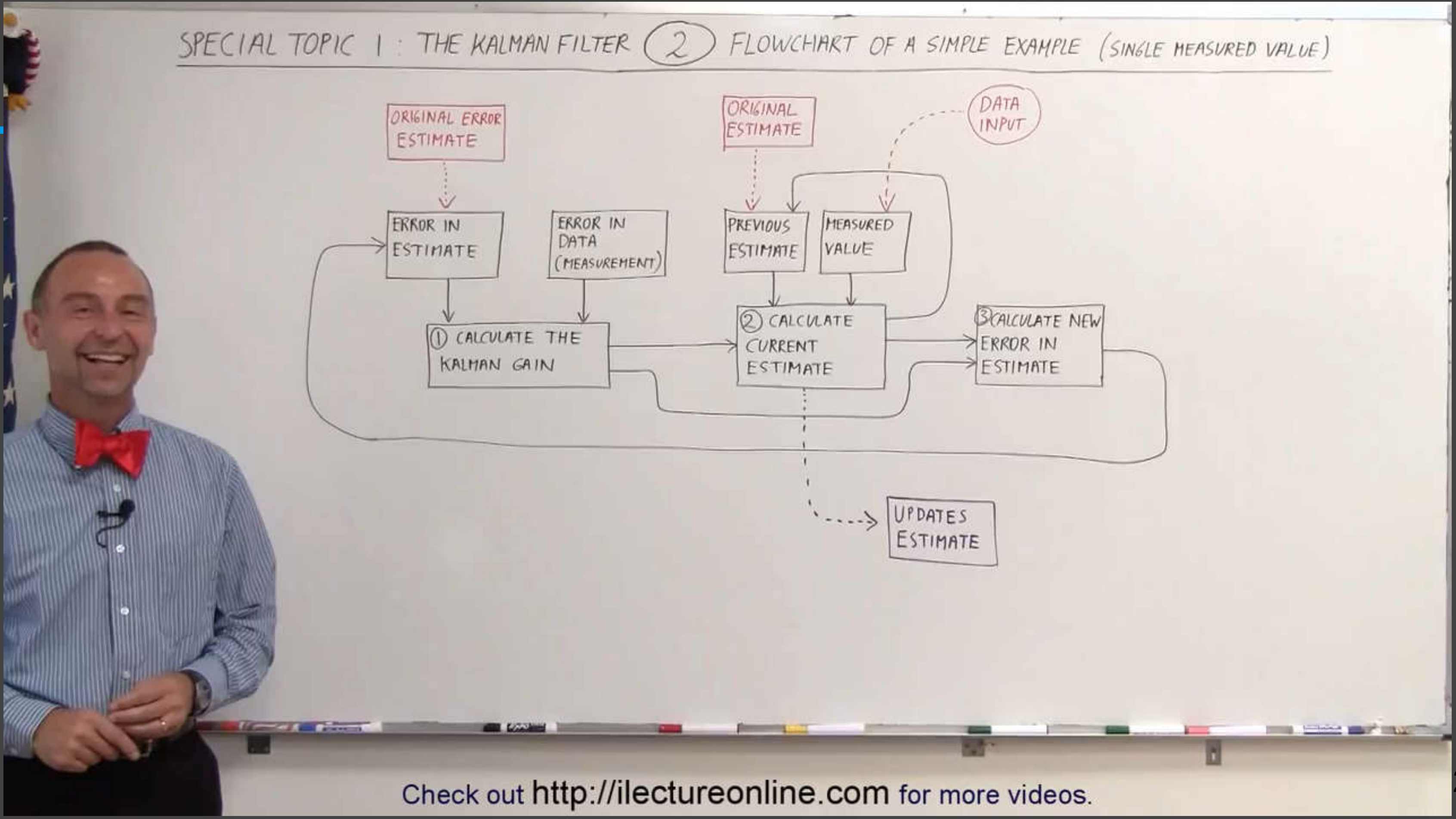
Kalman filtering is an algorithm that uses a series of measurements observed over time, containing statistical noise and other inaccuracies, and produces estimates of unknown variables.

Used for *computer vision, tracking, navigation, models of movement control.*

Kalman filter – 1



Kalman filter – 2

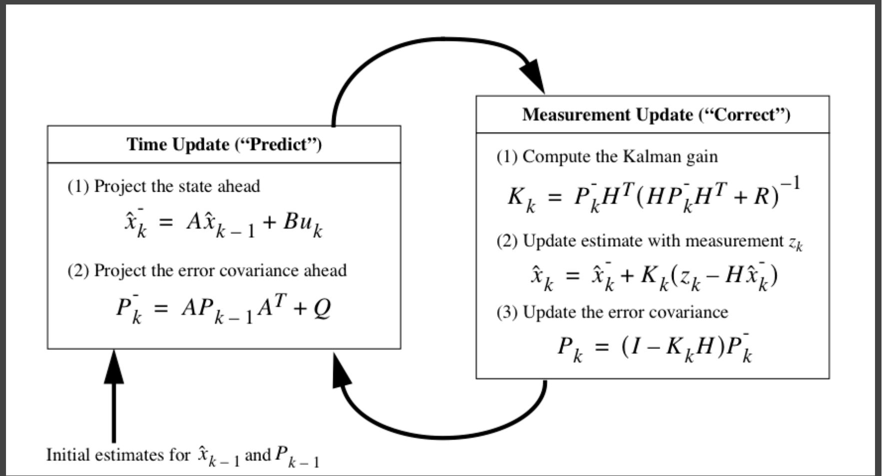


Operation of Kalman filter

The algorithm works in a two-step process.

In the prediction step, the Kalman filter produces estimates of the current state variables, along with their uncertainties.

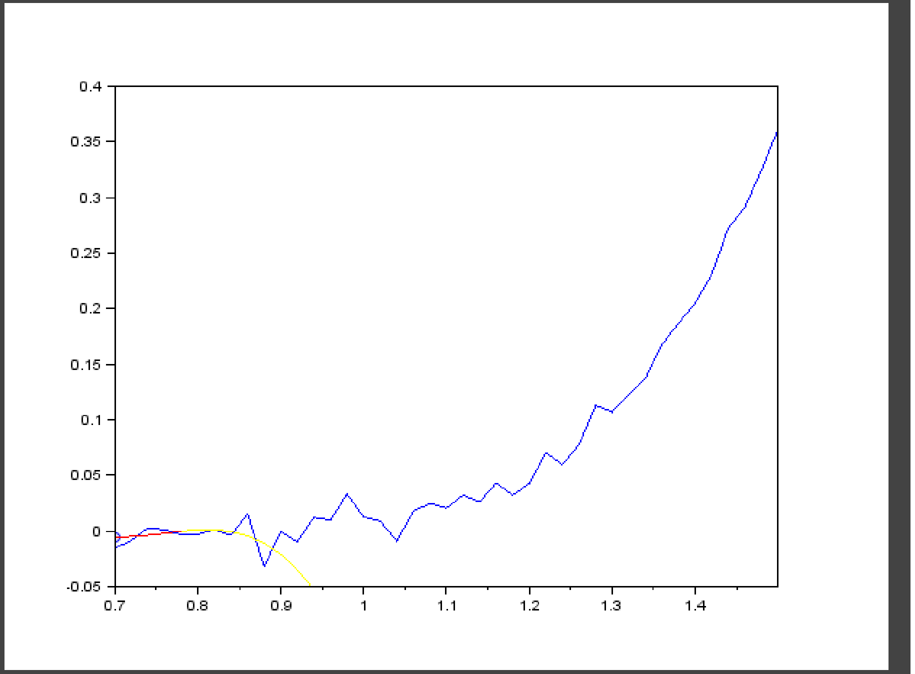
Once the outcome of the next measurement is observed, these estimates are updated using a weighted average, with more weight being given to estimates with higher certainty. The algorithm is recursive. It can run in real time.



Savitzky–Golay filter

Savitzky–Golay filter is a digital filter that can be applied to a set of digital data points for the purpose of smoothing the data, that is, to increase the signal-to-noise ratio without greatly distorting the signal.

This is achieved by fitting successive sub-sets of adjacent data points with a low-degree polynomial by the method of linear least squares.



Adaptive filter

Adaptive filtering involves the changing of linear filter parameters (coefficients) over time, to adapt to changing signal characteristics.

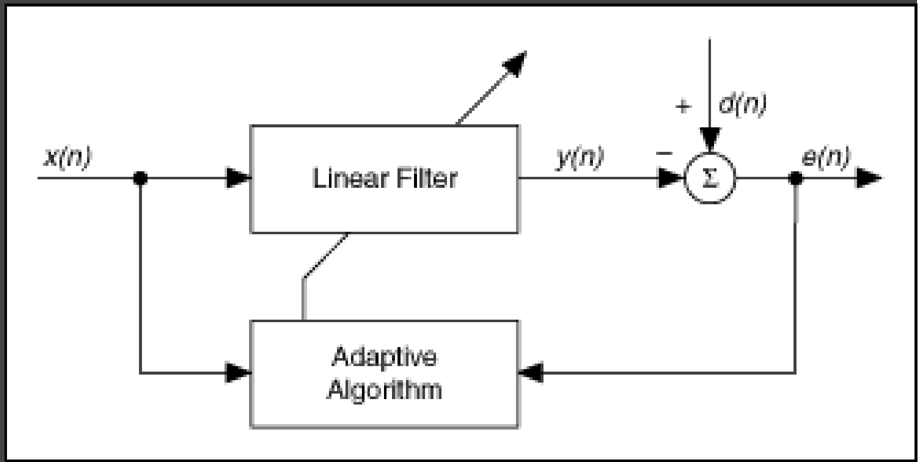
Adaptive filters can complete some signal processing tasks that traditional digital filters cannot:

- Remove noise whose power spectrum changes over time,
- Signal prediction,
- Adaptive feedback cancellation,
- Echo cancellation,
- Online system identification.

Adaptive Filtration

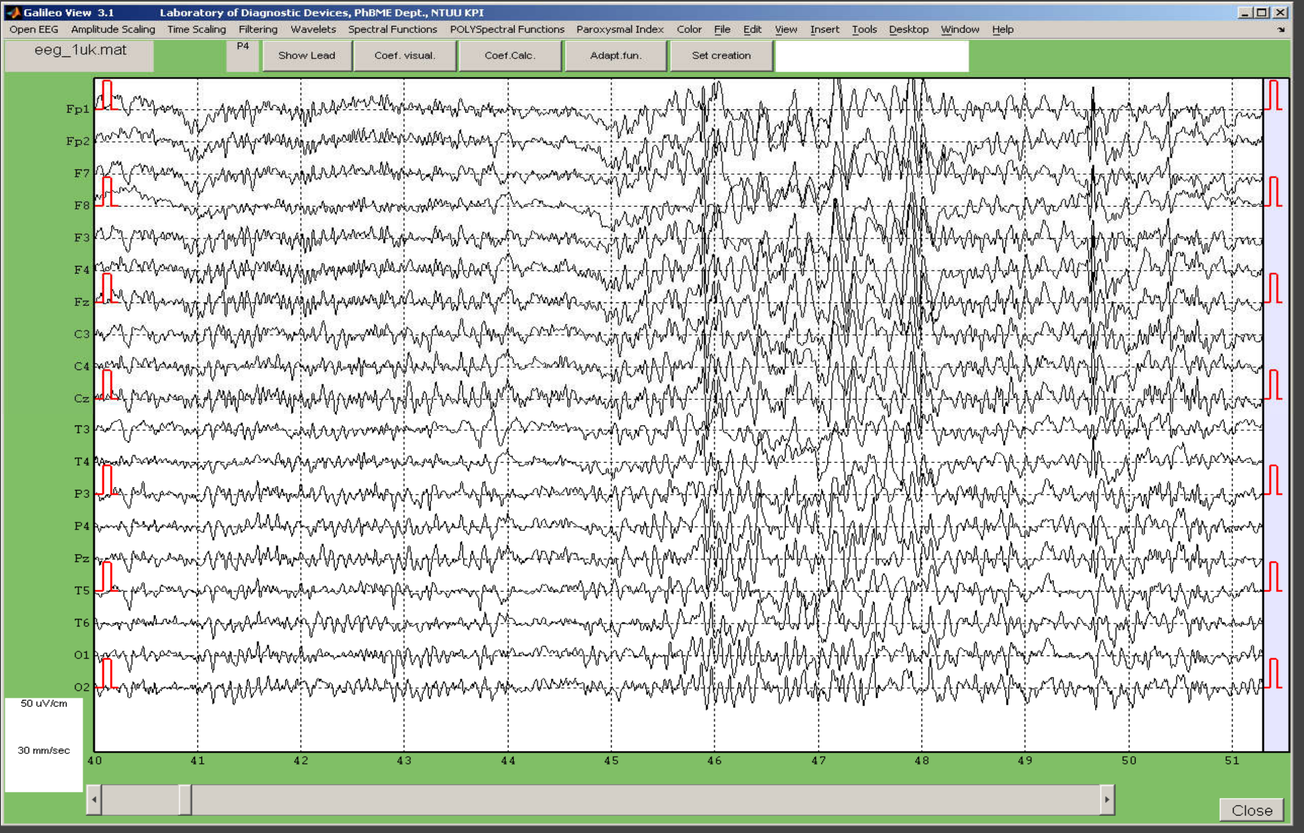
- x(n) is the input signal to a linear filter at time n
- y(n) is the corresponding output signal
- d(n) is an additional input signal to the adaptive filter
- e(n) is the error signal that denotes the difference between d(n) and y(n)

An adaptive algorithm adjusts the coefficients of the linear filter iteratively to minimize the power of e(n). For different applications, you choose the input and output signals x(n), d(n), y(n), and e(n) in different ways.



Waveform shape detection

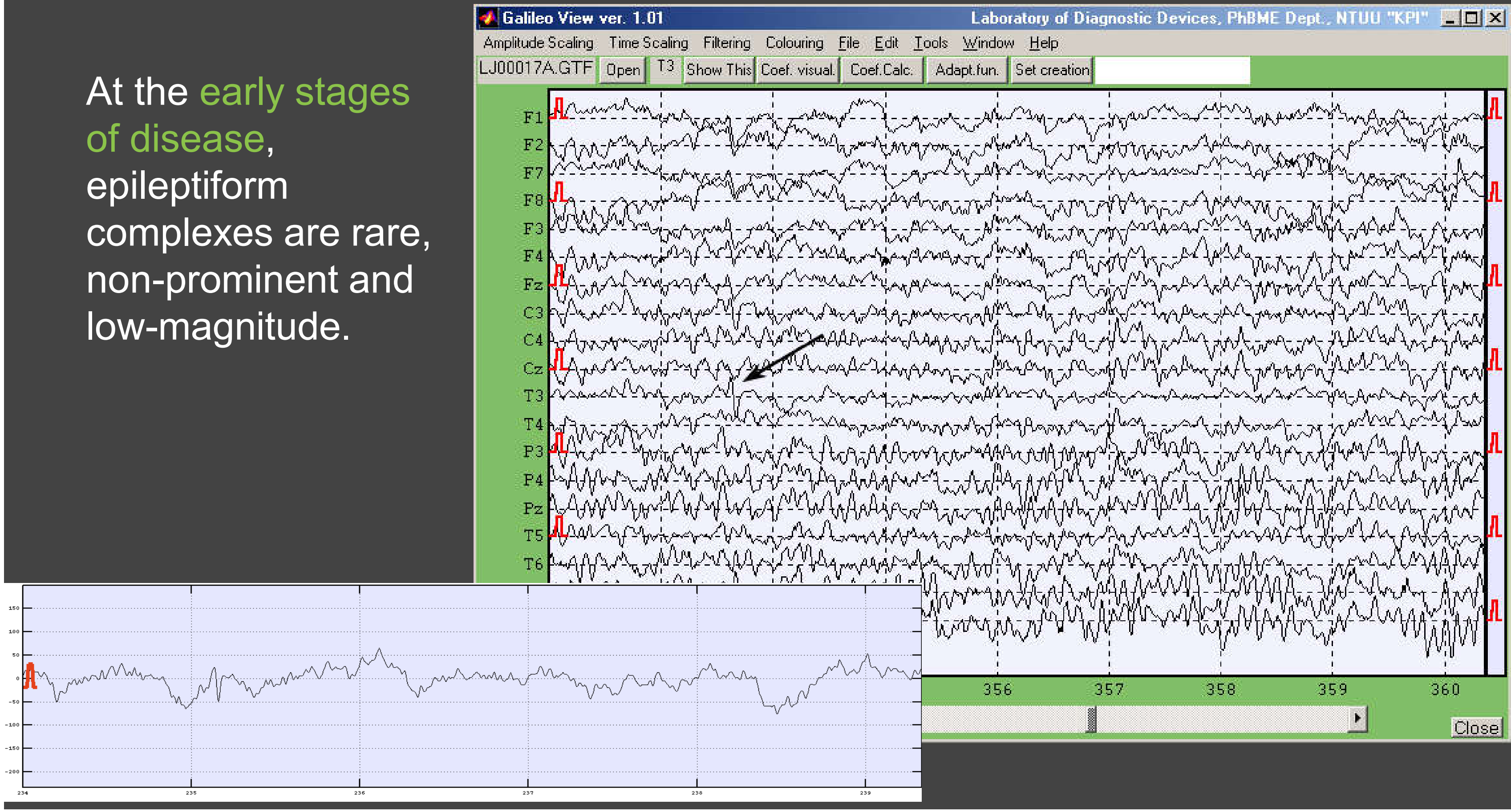
EEG during epileptic seizure




Galileo View ver. 1.02
Laboratory of Diagnostic Devices, PhBME Dept., NTUU "KPI"
Amplitude Scaling Time Scaling Filtering Wavelets Spectral Functions POLYSpectral Functions Color Paroxysmal Index File Edit Tools Window Help
Open
0
Show This
Coef. visual.
Coef.Calc.
Adapt.fun.
Set creation
F1
F2
F7
F8
F3
F4
Fz
C3
C4
Cz
T3
T4
P3
P4
Pz
T5
T6
O1
O2
600 uV/cm
30 mm/sec
40 41 42 43 44 45 46 47 48 49 50 51
Close

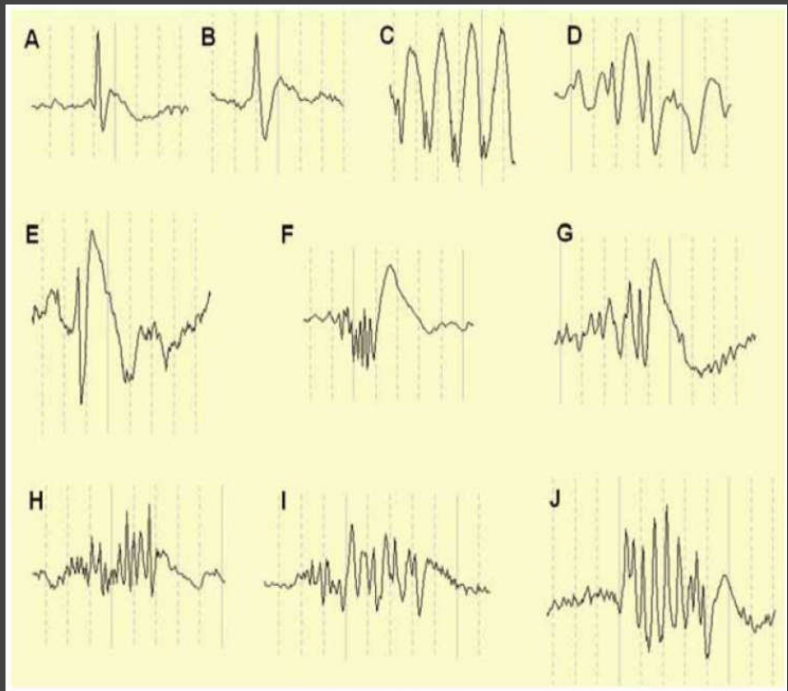
EEG of epileptic subject – rare complexes

At the early stages of disease, epileptiform complexes are rare, non-prominent and low-magnitude.



Epileptic Patterns

Epileptic oscillation complexes in human electroencephalogram - complexes of a sharp wave and a slow wave, indicating the presence of neuronal epileptic discharges in the brain.



Pattern Recognition by Template Matching

Template for the Class of Epileptiform Complexes

«generic» epileptiform complex

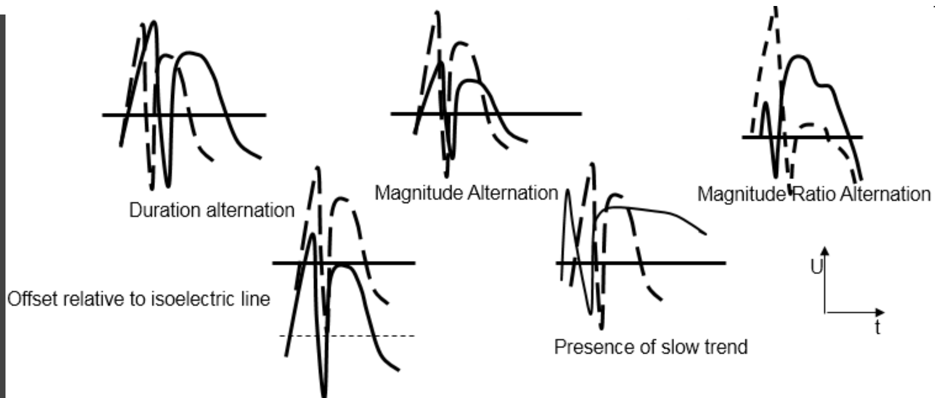
Parameters of possible acceptable distortions

Template for the class:

$$\{\eta(n)\}_{K_1} = \left\langle C_{K_1} + S_{K_1} + A_{K_1} \cdot \eta\left(L_{K_1}\right) \ , \ V_{K_1} \right\rangle$$

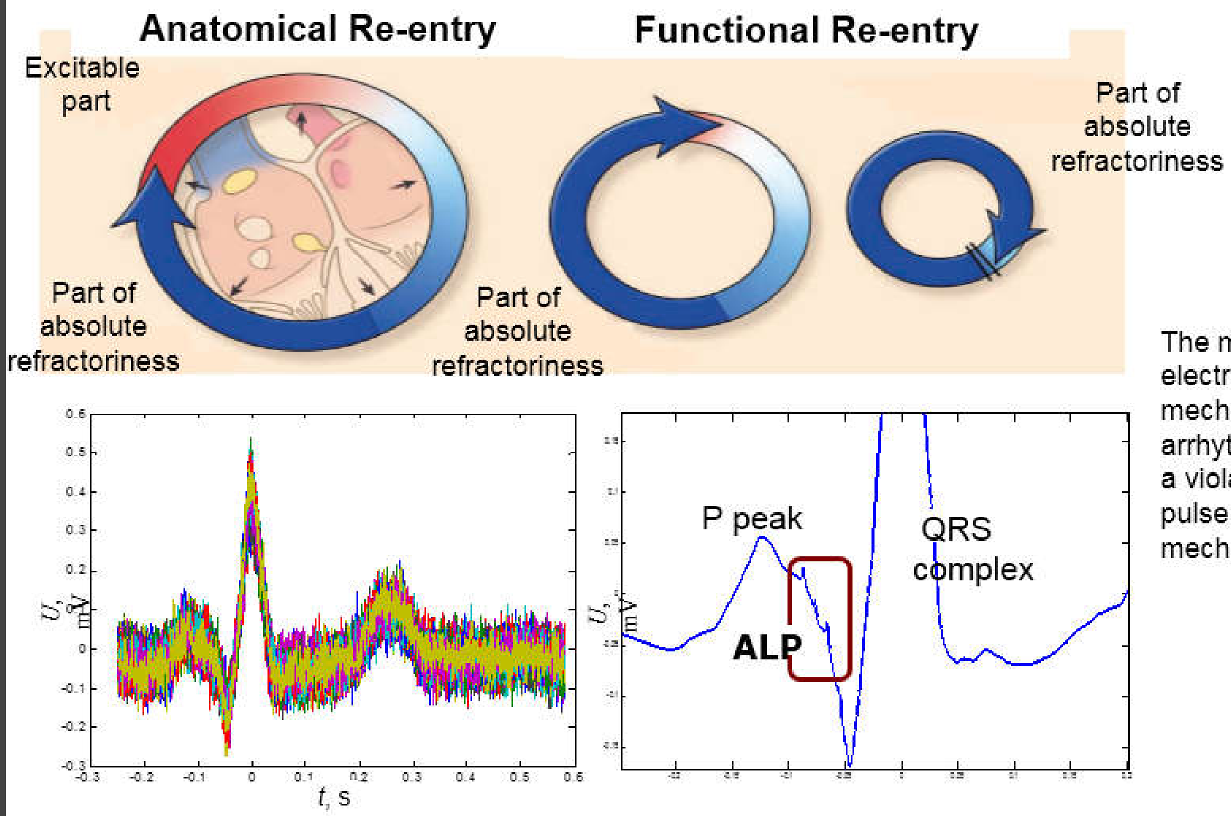
Pseudometric:

$$d_A\left(\eta_{K_1}, f\right) = \min_{i=1,Q}\left(\max_{n=1,z}\left|\eta(n) - f(n)\right|\right)$$



High-resolution ECG accumulation

Athrial late potentials in ECG as early signs of arrhythmias

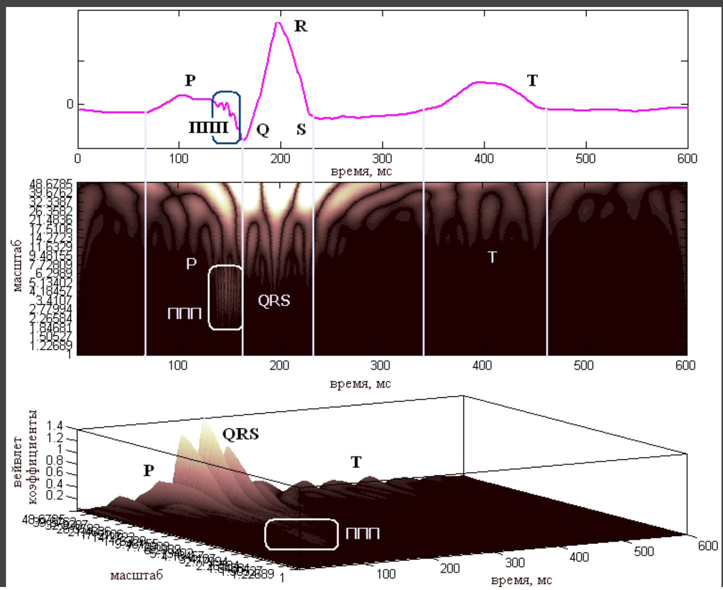


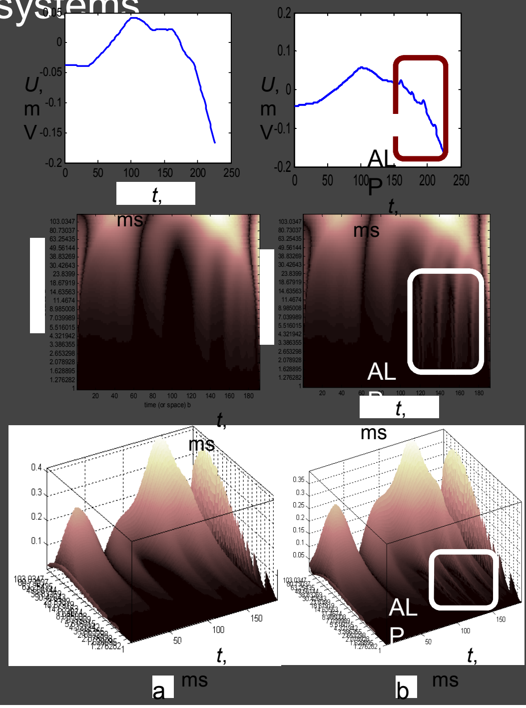
Aggregation of QRS complexes

The main electrophysiological mechanism leading to arrhythmia is caused by a violation of excitation pulse conducting, is the mechanism of re-entry.

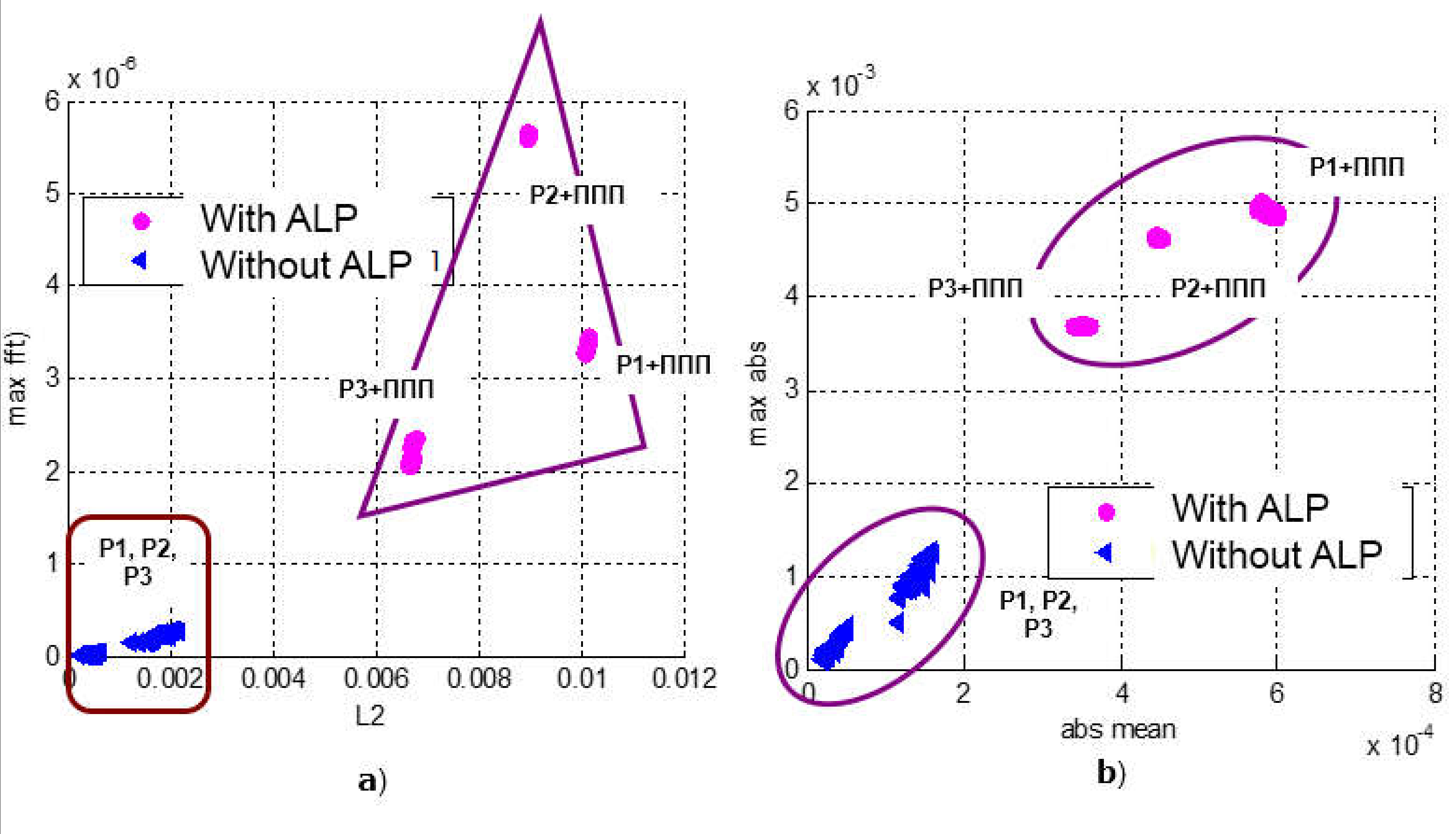
Noninvasive detection of ALP by wavelet transform

Investigation of possibilities of joint use of wavelet transforms and expansions in the basis of eigenvectors, as well as the development of algorithms for pattern recognition in ALP detection will improve diagnostic accuracy in modern ECG systems.





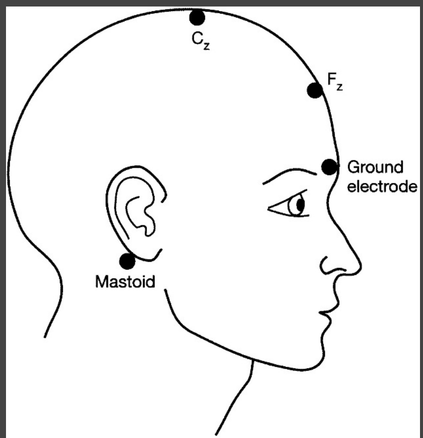
Athrial Late Potentials in ECG classification



Auditory evoked response averaging

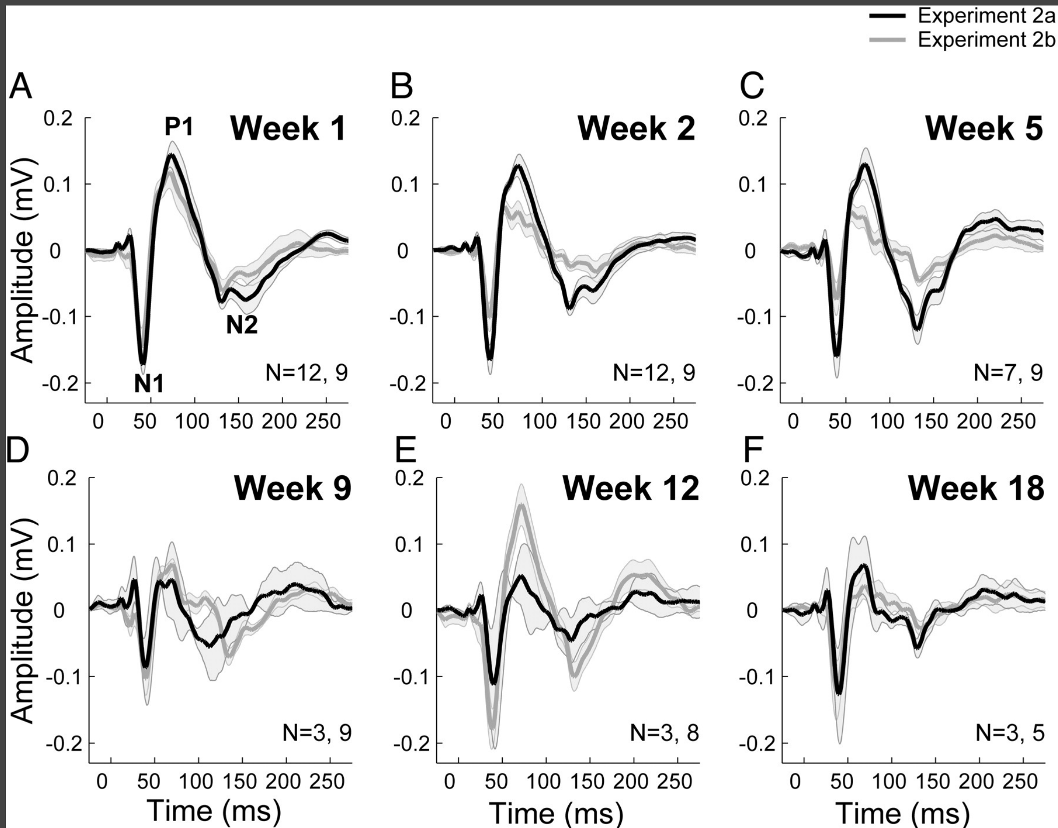
EEG auditory evoked potentials

Auditory brainstem response (ABR) is a neurologic test of auditory brainstem function in response to auditory (click) stimuli.
It's a set of seven positive waves recorded during the first 10 seconds after a click stimuli. They are labeled as I - VII



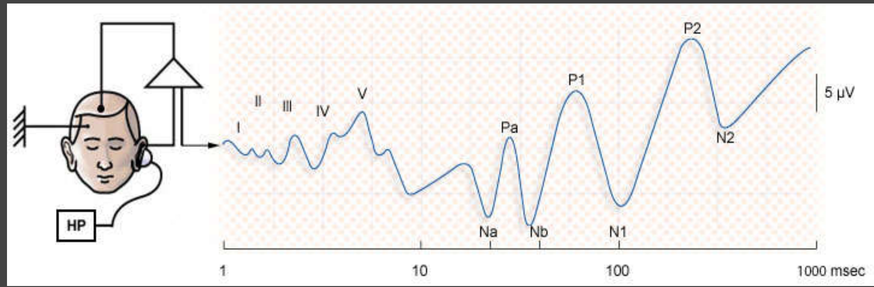
- Cz (at vertex) (recording electrode)
- Ipsilateral ear lobule or mastoid process (reference electrode)
- Contra lateral ear lobule (act as a ground)

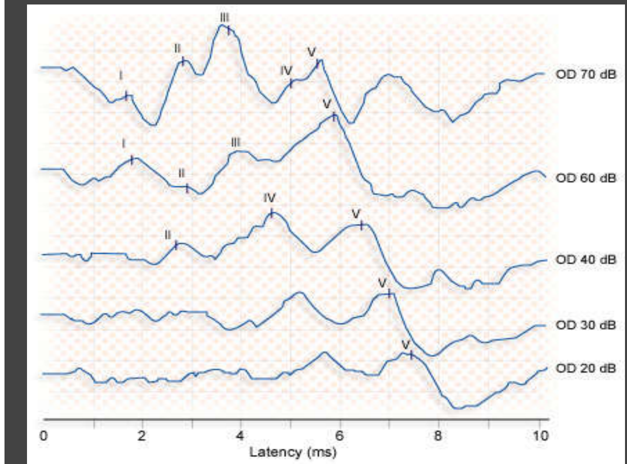
AEP averaging

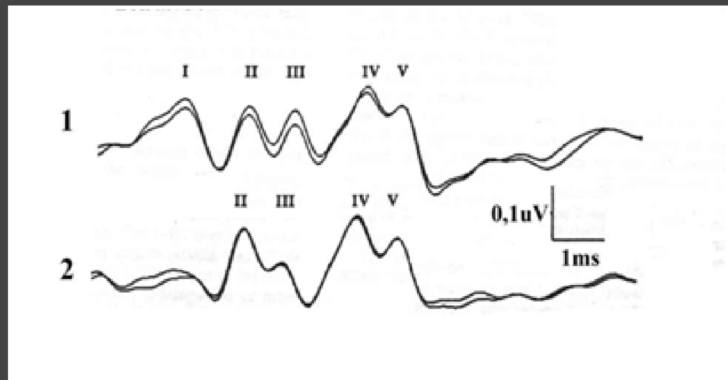


AEP signal parameters

There are short-, medium- and long-latency AEP but most often hearing diagnosed with the **medium-latency AEP**.



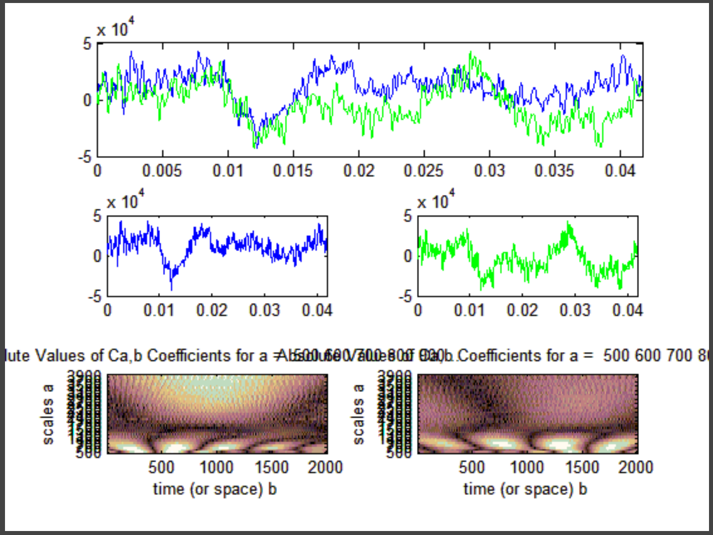


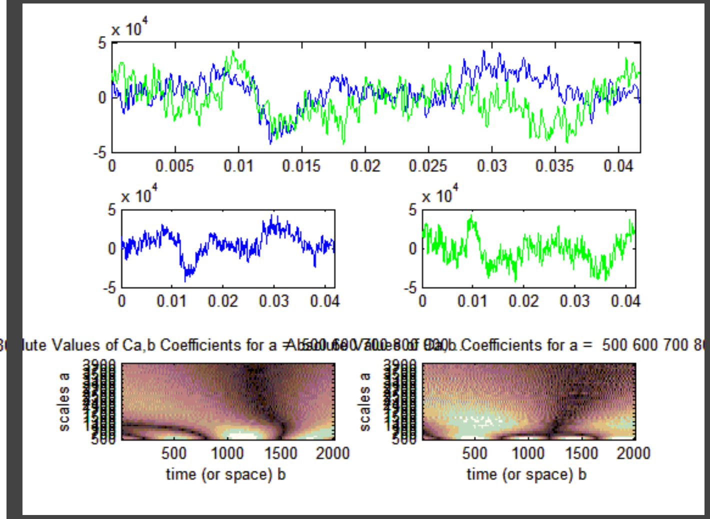


	I	II	III	IV	V
Latency (ms)	1,6-1,8	2,7-3,0	3,7-4,1	4,9-5,3	5,7-6,0

Techniques of AEP signal analysis

Wavelet transform





Methods based on continuous wavelet transform (CWT) are used, which can be applied to detection of V-wave in noise signal.

Sensitivity – 83 %

Selectivity – 87 %

Specificity – 75 %