

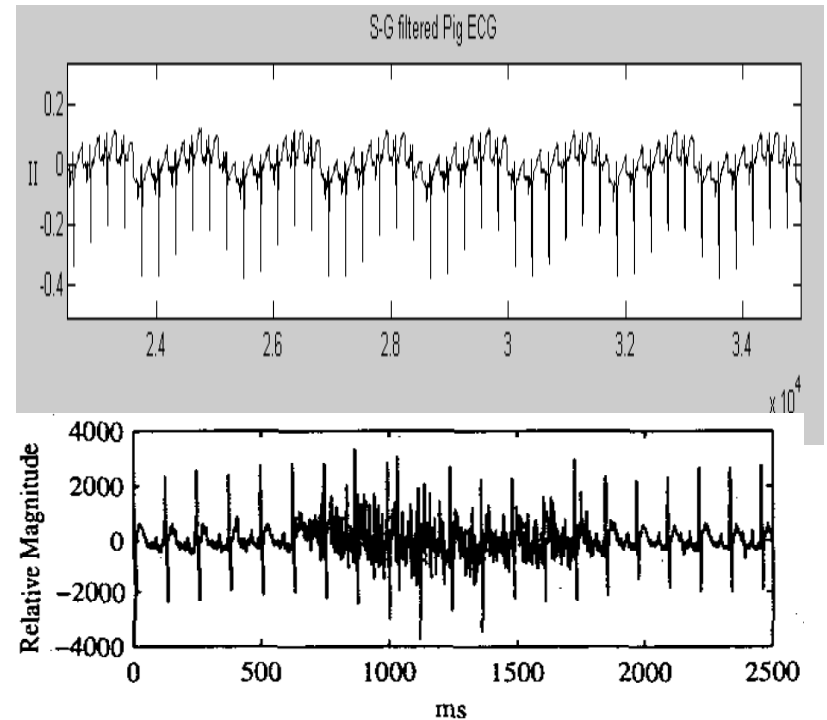
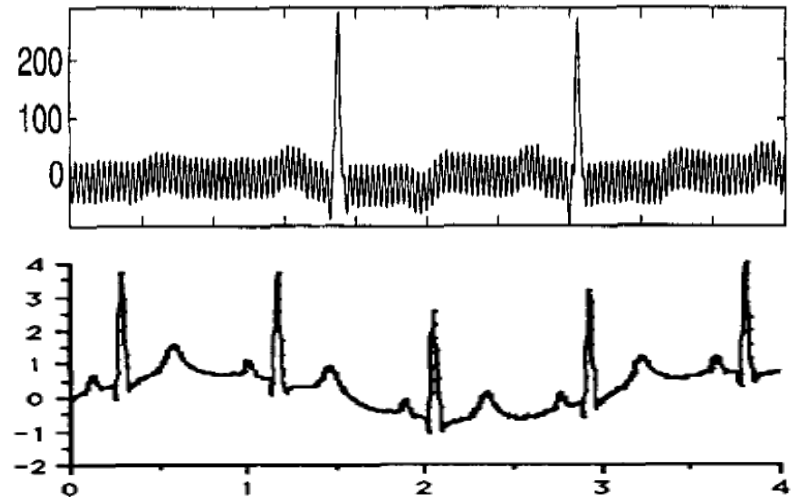
# Biosignal filtering and artifact rejection

Biosignal processing I, 521273S  
Autumn 2019

# Motivation

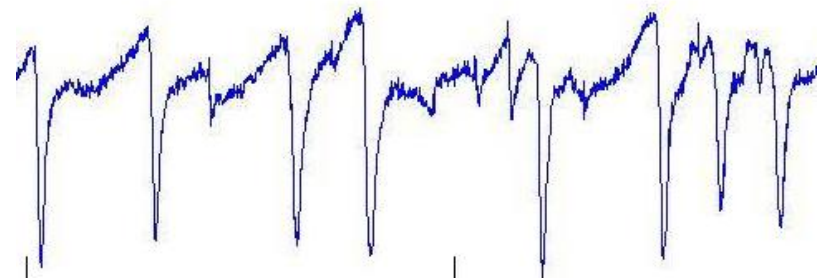
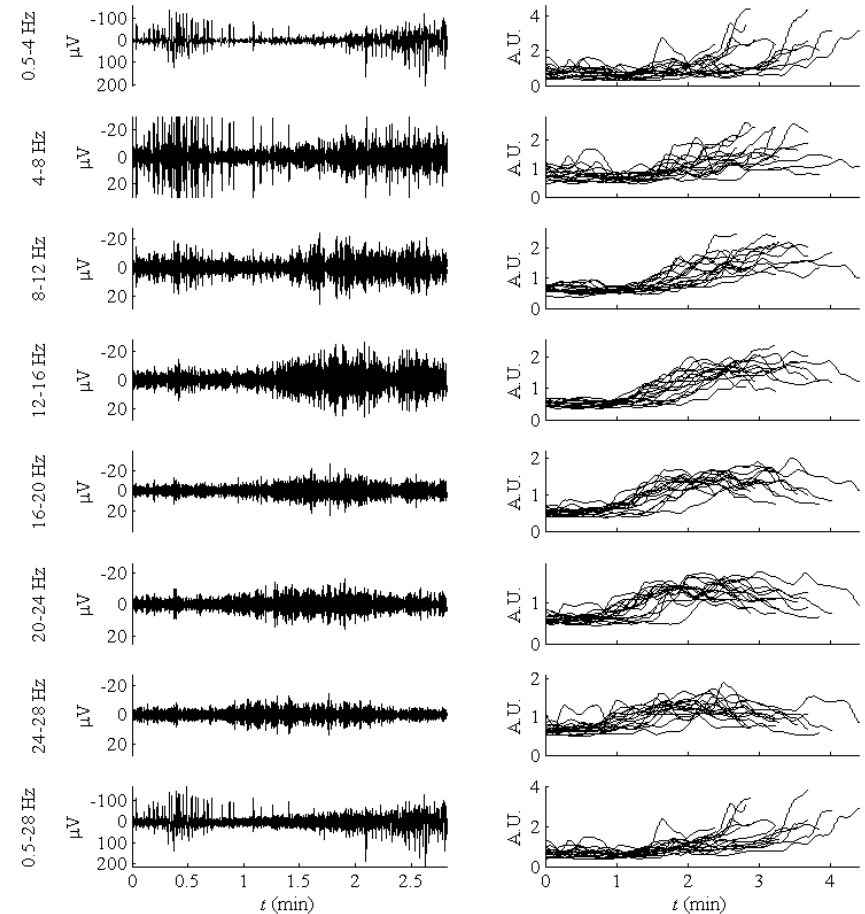
## 1) Artifact removal

- power line
- non-stationarity due to baseline variation
- muscle or eye movement artifacts in EEG or ECG
- Solution?: epoch rejection due to artifacts



## 2) Enhancement of useful information

- bandpass filtering
- finding certain signal waveforms such as eye blinks from EEG or QRS complexes from ECG
- smoothing for illustrative purposes



A few words on an example  
signal: ECG

# ECG:Electrocardiogram

- Electrical potential changes due to contractile activity of the heart
- Measured usually by standard 12-lead system
  - With four limb electrodes and six chest electrodes
- Common ECG-applications are
  - stationary ECG
  - Holter-monitoring
  - stress-ECG (exercise testing)
  - telemedicine applications
  - heart rate monitors
- Invasive instrumentation:
  - heart pacemakers
  - arrhythmia-pacemakers

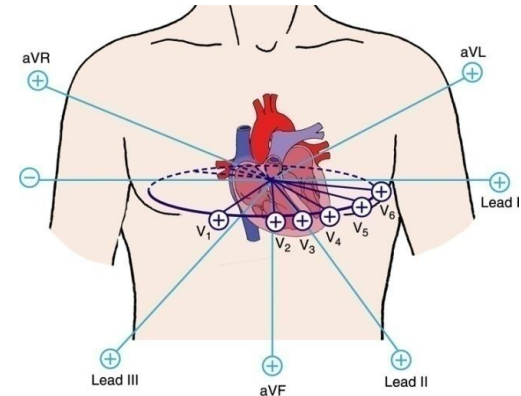
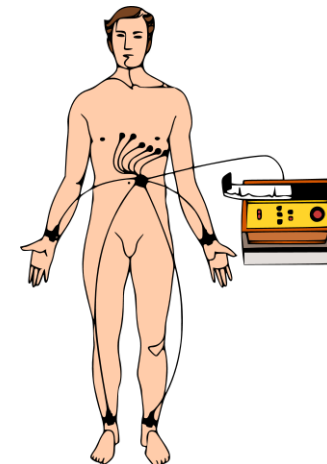


Figure 17-42 Electrocardiographic views of the heart.

Copyright © 2005 Lippincott Williams & Wilkins. Instructor's Resource CD-ROM to Accompany Critical Care Nursing: A Holistic Approach, eighth edition.

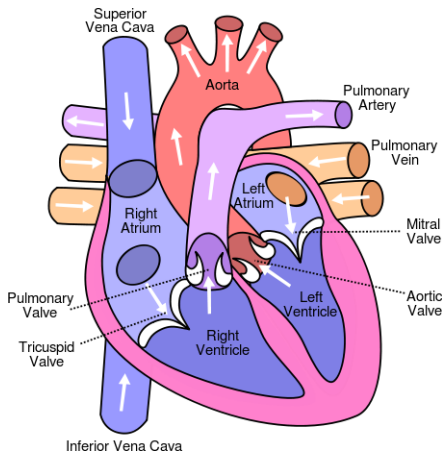
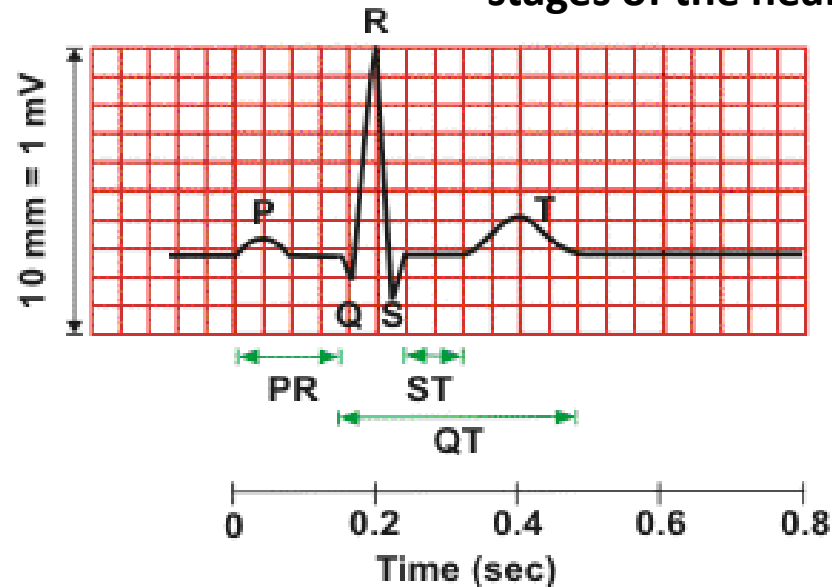


# ECG structure

## E.g. Feature analysis

Automatic detection of different segments and waves (amplitudes, intervals)

Contraction and relaxation stages of the heart



P wave (0.08 - 0.10 s)

QRS (0.06 - 0.10 s)

P-R interval (0.12 - 0.20 s)

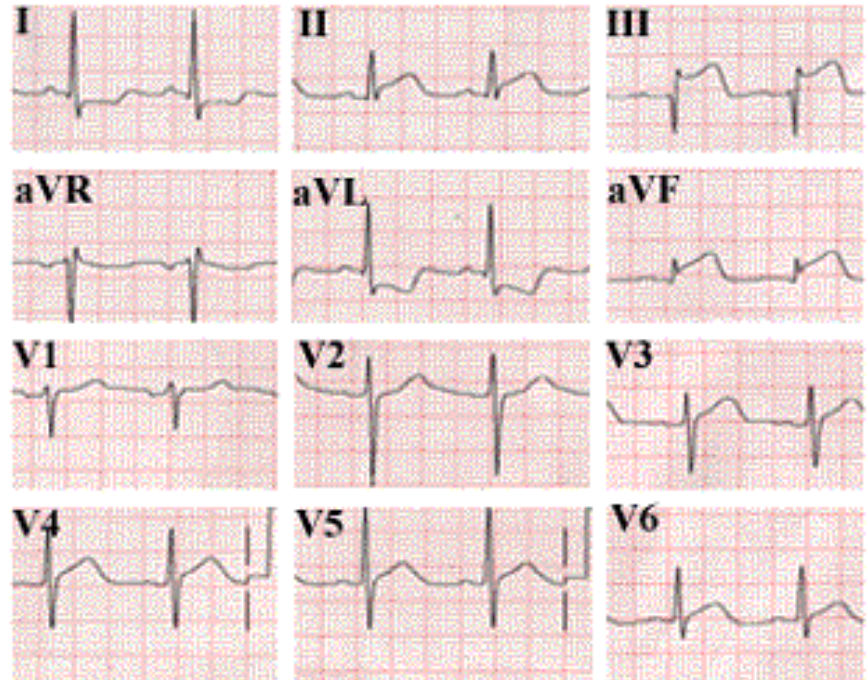
Q-T<sub>c</sub> interval (≤ 0.44 s)\*

$$*QT_c = QT / \sqrt{RR}$$

Systemic vs. pulmonary circulation

# 12-lead ECG

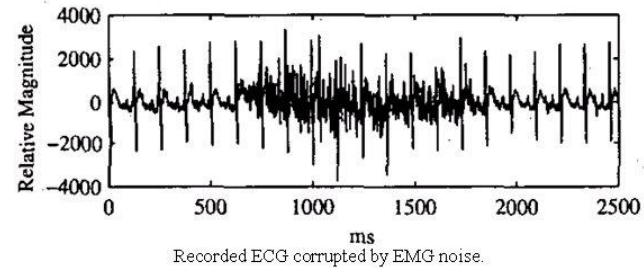
- ECG analysis focus:
  - QRS complex detection
  - feature analysis
  - classification of arrhythmias
  - ECG signal compression
  - Heart rate variability (HRV) analysis



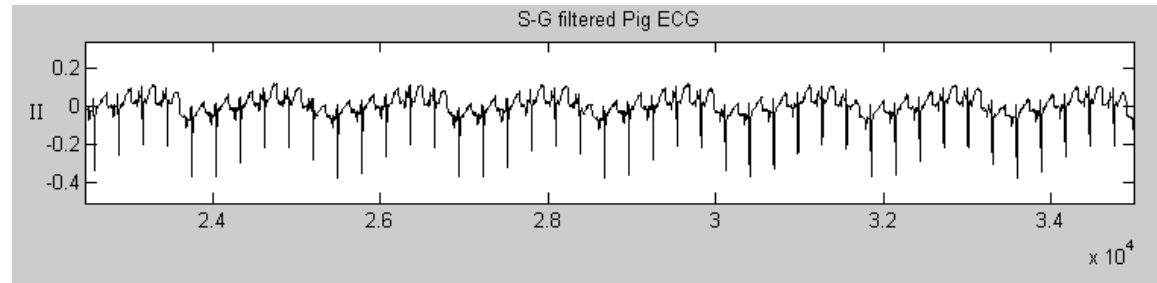
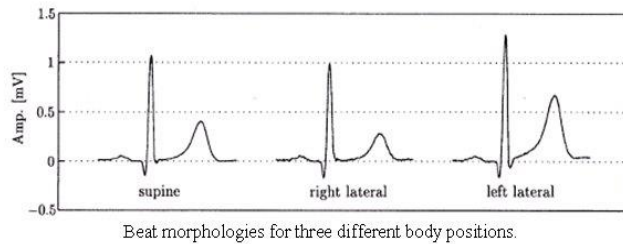


# Noise in ECG

Distortion caused by electromyogram:



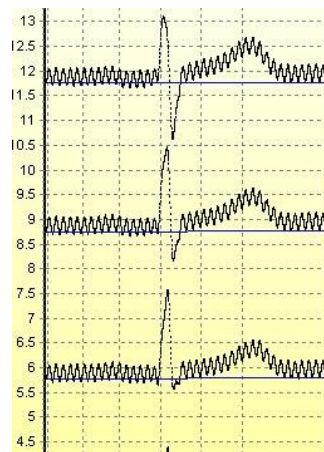
Effect of body position on ECG:



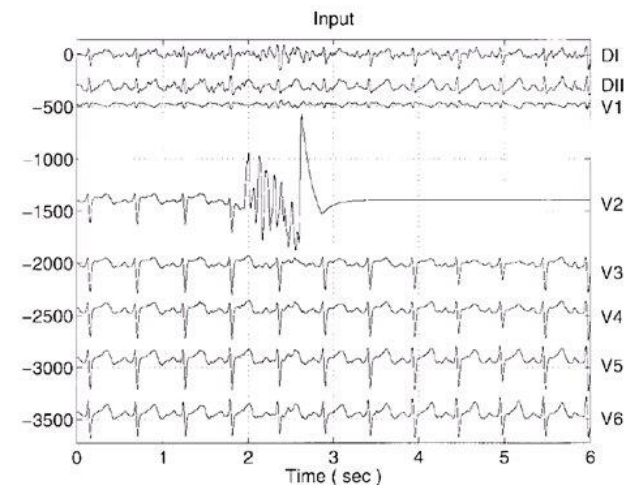
Respiration-induced ECG modulation:



ECG (upper trace) and respiration measured by a pneumatic respiration transducer placed around chest (lower trace).



Channel loss:



Complete loss of channel V<sub>2</sub> during recording.

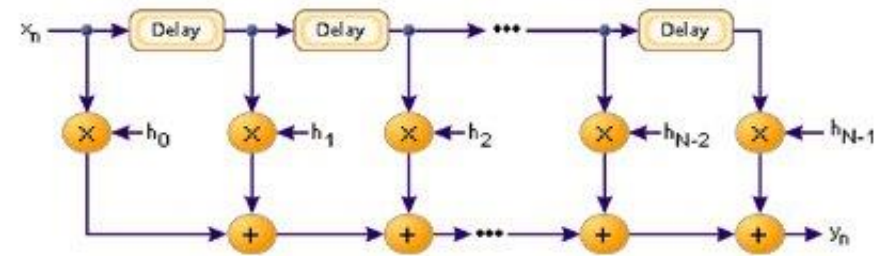


Figure 2. The logical structure of an FIR filter

# Basic filtering techniques

# FIR filters

- **Finite Impulse Response (FIR) filter**
  - Stable
  - Simple to implement
  - Linear phase response
    - Symmetrical impulse response
    - All frequencies have the same amount of delay – no phase distortion

$$y(n) = \sum_{k=0}^{N-1} h(k)x(n-k)$$

$$H(z) = \sum_{k=0}^{N-1} h(k)z^{-k}$$

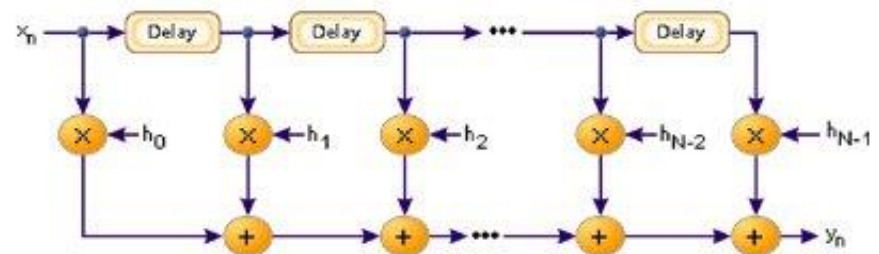


Figure 2. The logical structure of an FIR filter

# FIR

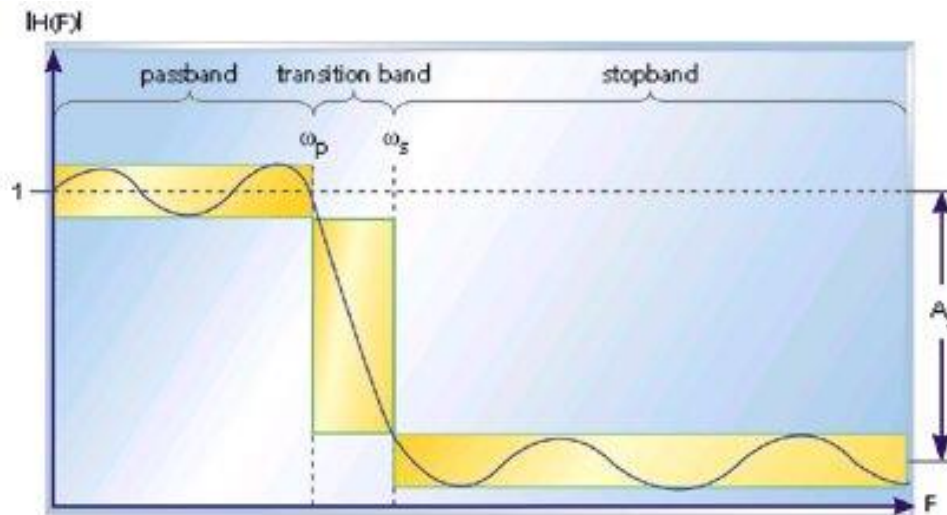
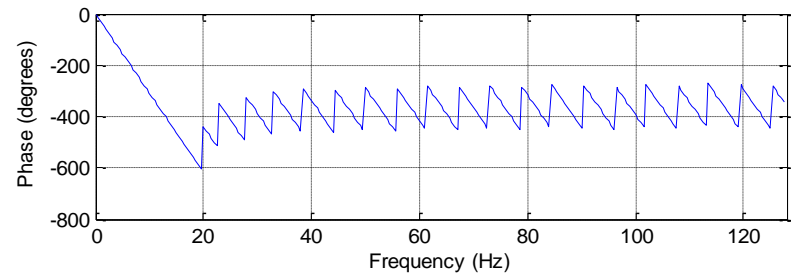
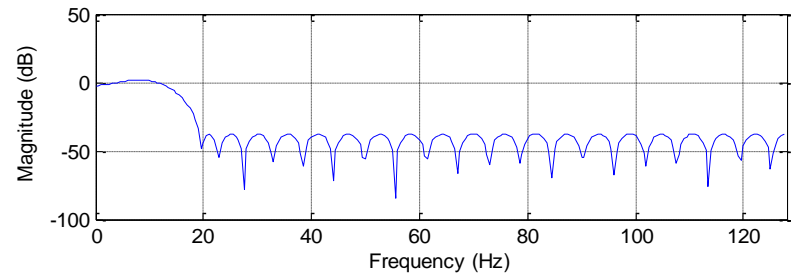
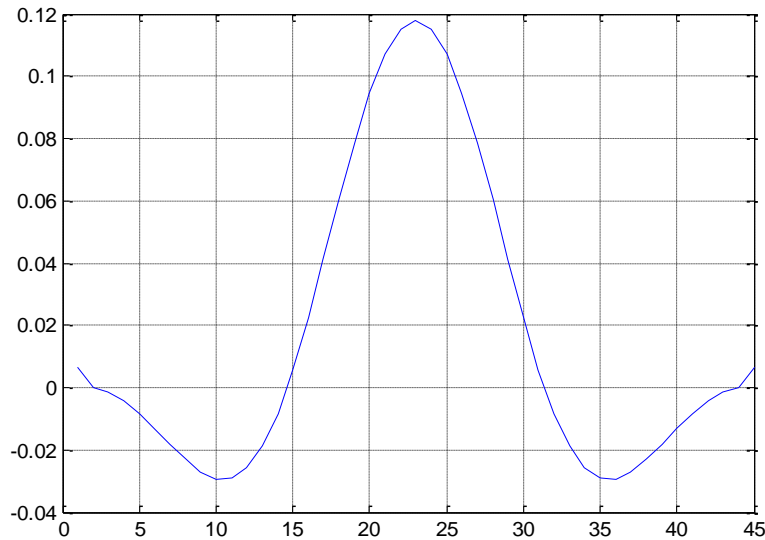
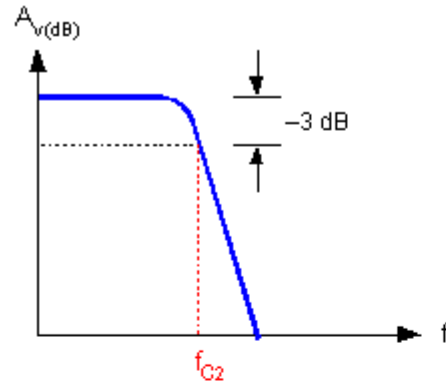


Figure 1. The response of a lowpass filter to various input frequencies

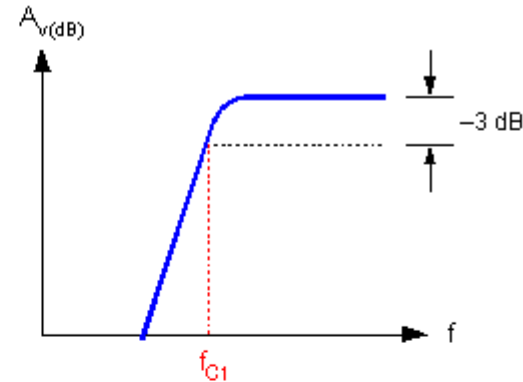
Lowpass filter, order 44 ( $N=45$ ),  
positive symmetry.  
 $f_s=256$  Hz,  $F_p=13$  Hz,  $F_s=19$  Hz,  
 $R_p=4$  dB,  $A_s=38$  dB

Filter characteristics:  
sampling frequency  $f_s$ ,  
passband  $F_p$ , stopband  $F_s$ ,  
ripple  $R_p$ , attenuation  $A_s$

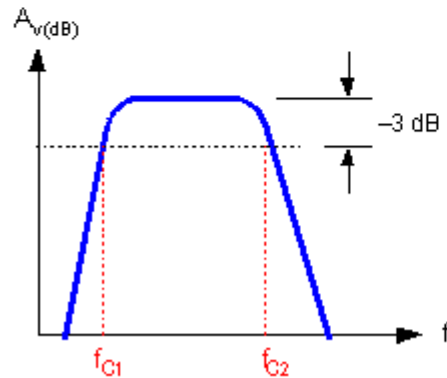
# Filter types by spectral characteristics



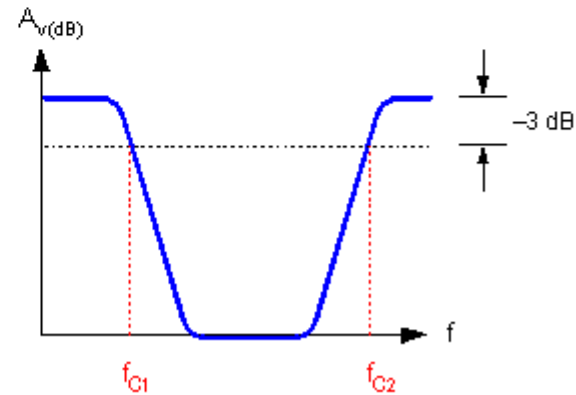
Low-pass filter



High-pass filter



Band-pass filter



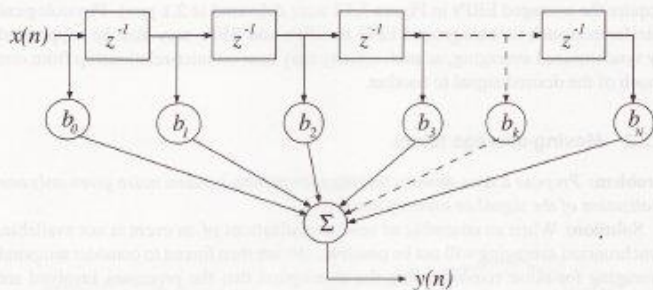
Band-stop (notch) filter

# Smoothing: averaging filter

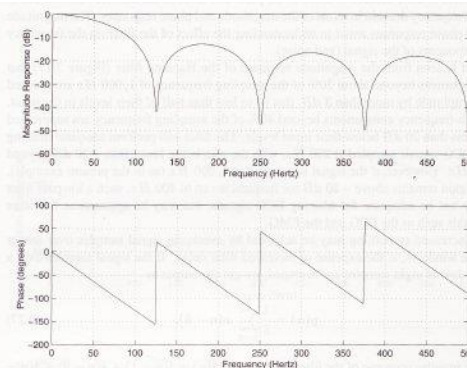
- Average of a sliding window of size N samples
  - FIR filter

$$y(n) = \frac{1}{N} \sum_{k=0}^{N-1} x(n-k) = \frac{1}{N} [x(n-0) + x(n-1) + \dots + x(n-N+1)]$$

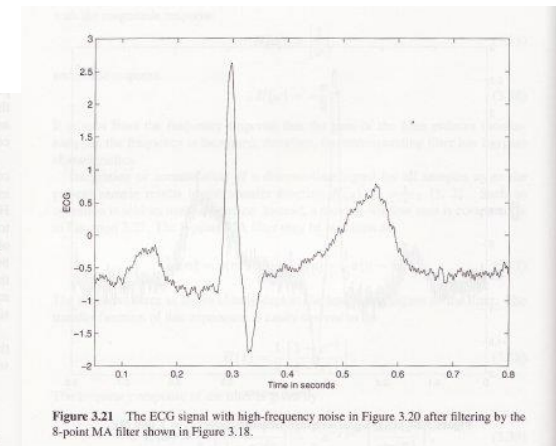
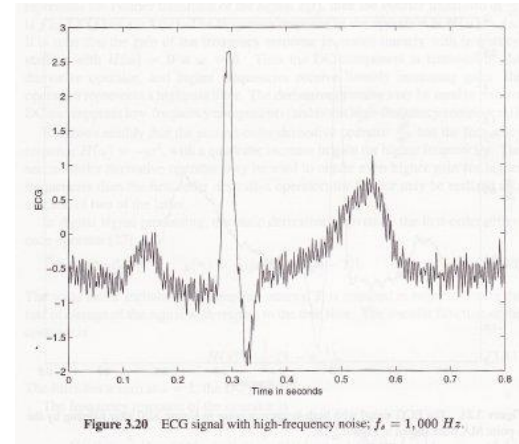
$$= \frac{1}{N} x(n-0) + \frac{1}{N} x(n-1) + \dots + \frac{1}{N} x(n-N+1)$$



**Figure 3.15** Signal-flow diagram of a moving-average filter of order  $N$ . Each block with the symbol  $z^{-1}$  represents a delay of one sample, and serves as a memory unit for the corresponding signal sample value.



**Figure 3.18** Magnitude and phase responses of the 8-point moving-average (smoothing) filter.



# Smoothing: Hanning filter

$$H(z) = \frac{1}{4}[1 + 2z^{-1} + z^{-2}]$$

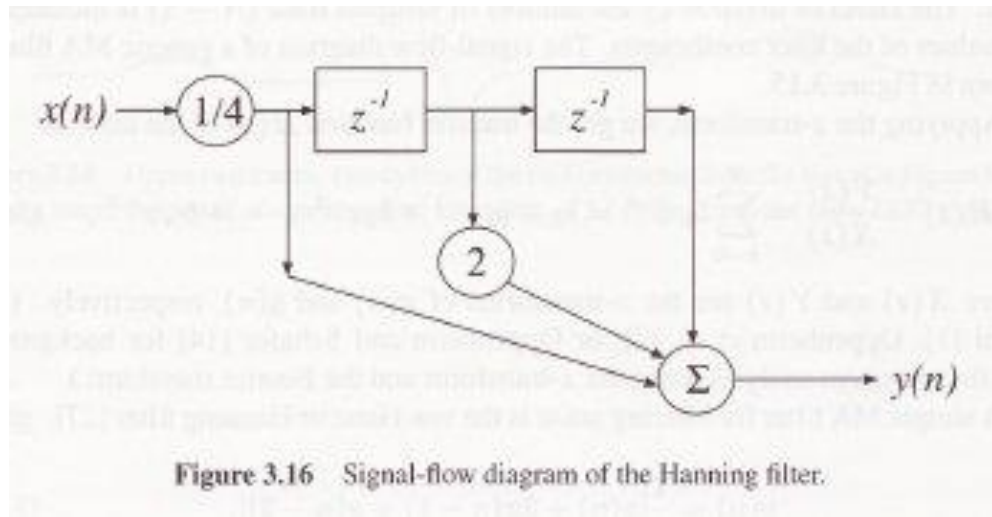


Figure 3.16 Signal-flow diagram of the Hanning filter.

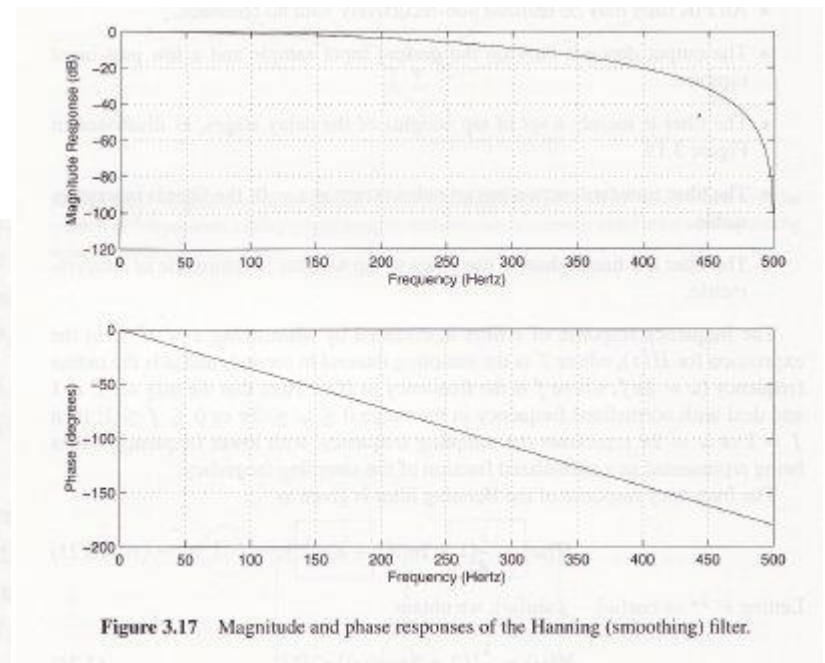


Figure 3.17 Magnitude and phase responses of the Hanning (smoothing) filter.

# IIR filters

- Infinite Impulse Response (IIR) filters

- Feedback system
- Normally fewer coefficients than with FIR
- Used for sharp cut-off (notch filters for example)
- Can become unstable or performance degrade if not designed with care
- Pole-zero diagram
- Nonlinear phase characteristics causes phase distortion altering harmonic relationships – frequency components have different time delays (often undesirable)
  - The wave shapes are distorted!

$$y(n) = \sum_{k=0}^{\infty} h(k)x(n-k) = \sum_{k=0}^N b_k x(n-k) - \sum_{k=1}^M a_k y(n-k)$$

$$H(z) = \frac{b_0 + b_1 z^{-1} + \dots + b_N z^{-N}}{1 + a_1 z^{-1} + \dots + a_M z^{-M}} = \frac{\sum_{k=0}^N b_k z^{-k}}{1 + \sum_{k=1}^M a_k z^{-k}}$$

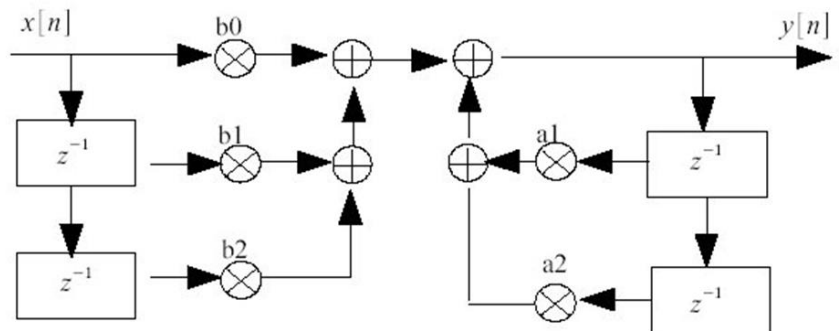


Figure 6: Network diagram for an IIR filter having two poles and two zeros.

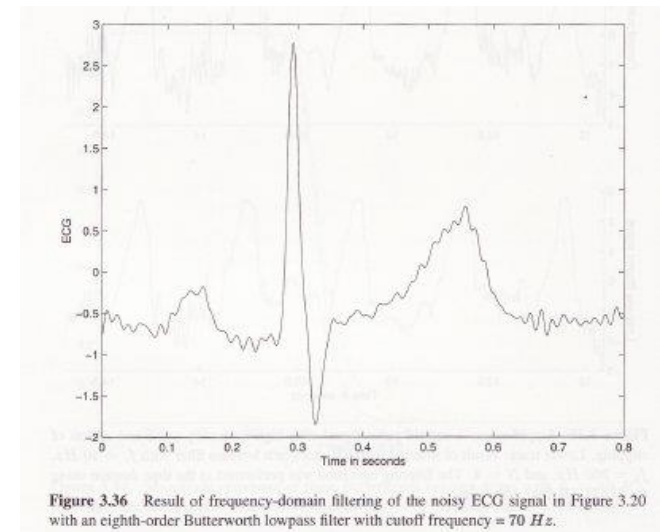
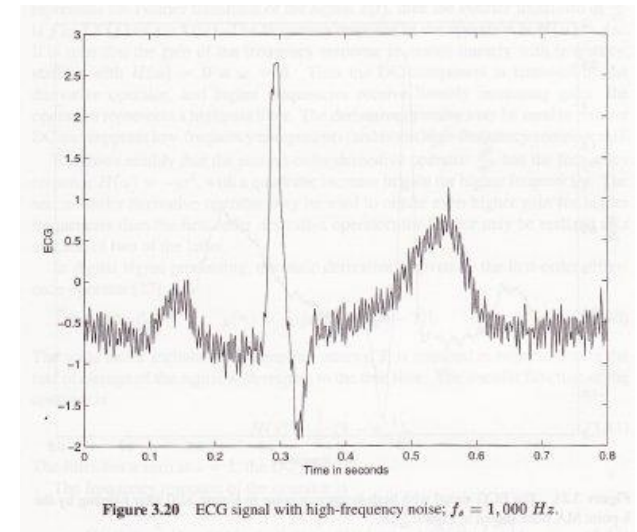
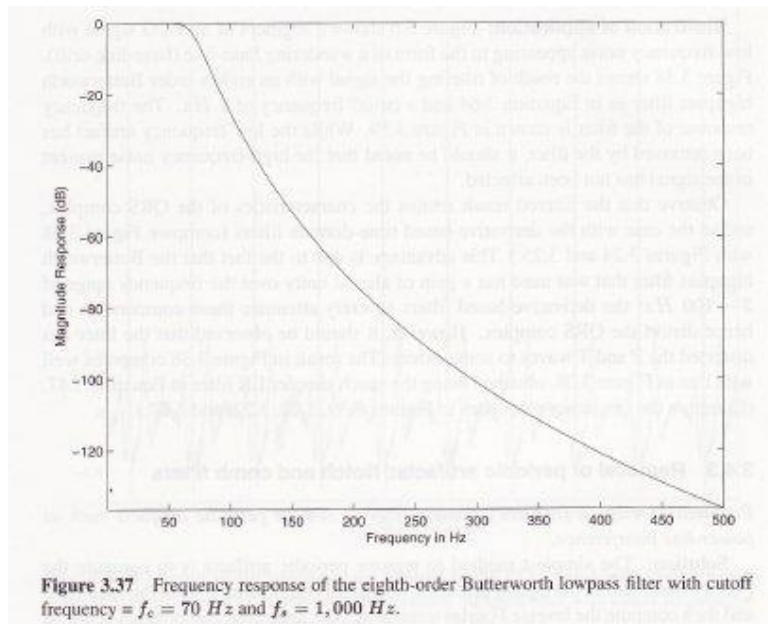
Source: <http://www.triplecorrelation.com/courses/fundsp/iioverview.pdf>



# Smoothing: Butterworth lowpass filtering

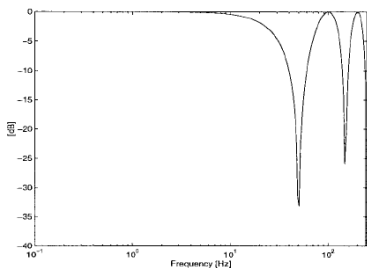
## Butterworth lowpass filter

- Select suitable order and cutoff frequency
- Maximally flat magnitude filter



# Notch/comb filter

- Often used for 50/60 Hz power line artifact filtering
- Narrow stop-band in basic and harmonic frequencies
- Be careful with the aliased harmonics
- Can be implemented as FIR or IIR



$$y(t) = x(t) - 2 \cdot x(t - 16) + x(t - 32) \\ - 2 \cdot y(t - 8) - y(t - 16).$$

Ruha et al. (1997)

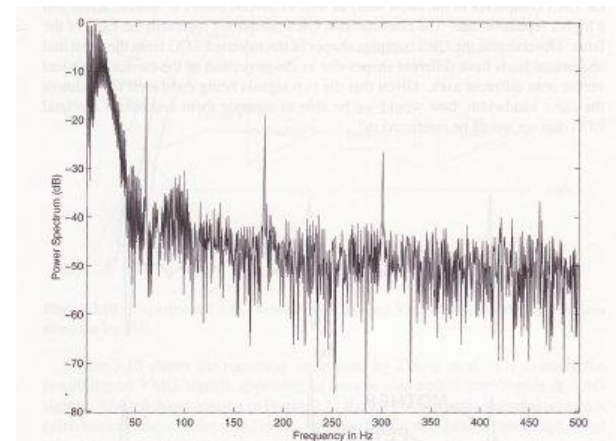
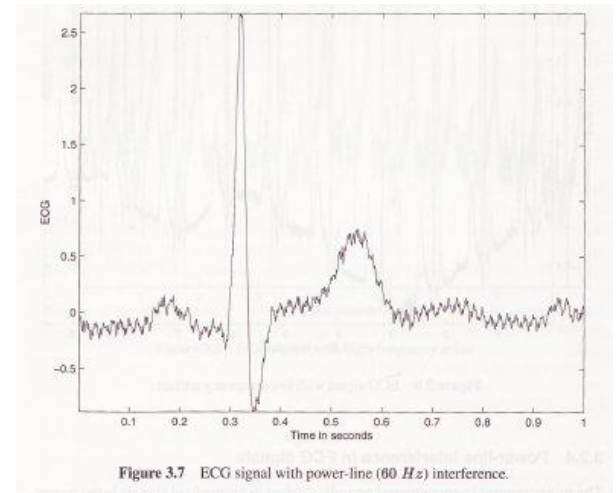
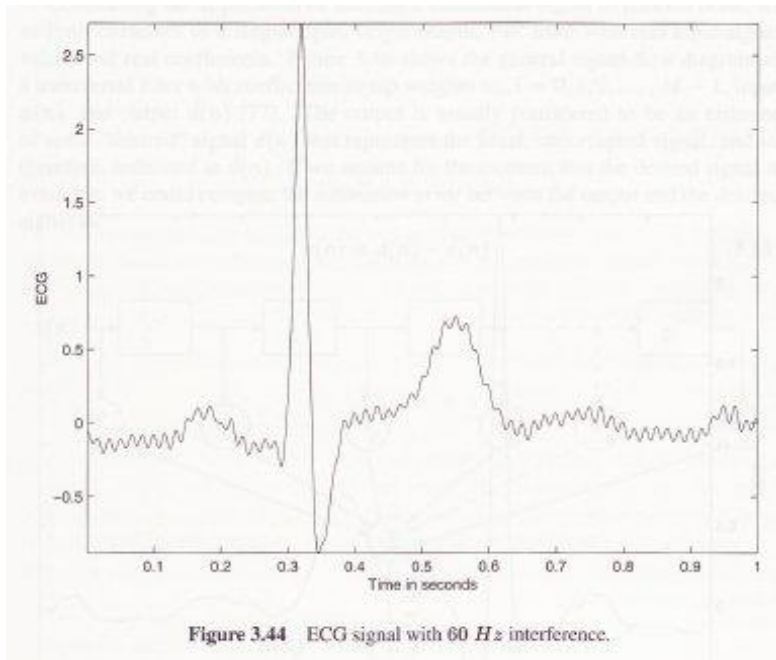
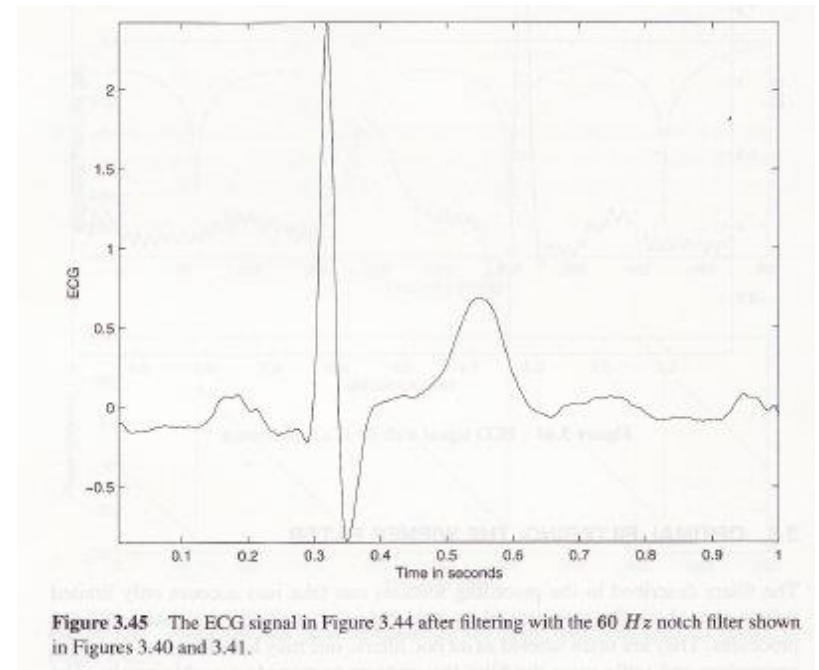


Figure 3.8 Power spectrum of the ECG signal in Figure 3.7 with power-line interference. The spectrum illustrates peaks at the fundamental frequency of 60 Hz as well as the third and fifth harmonics at 180 Hz and 300 Hz, respectively.

# Notch/comb filter

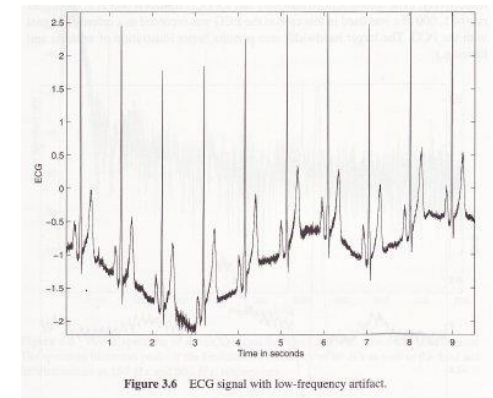


Original signal



Filtering result

# Trend removal



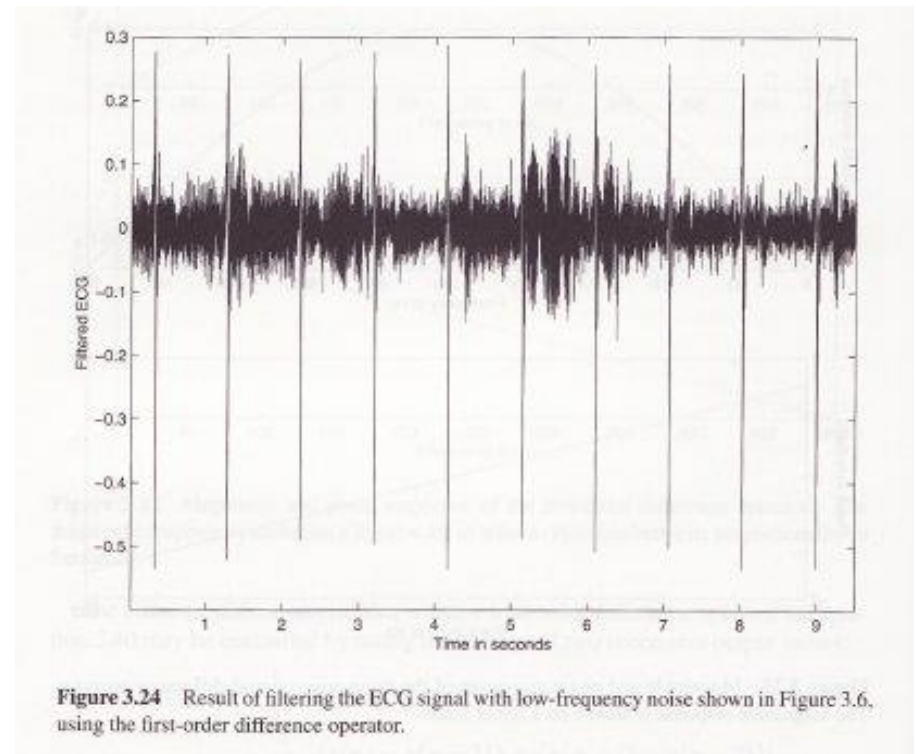
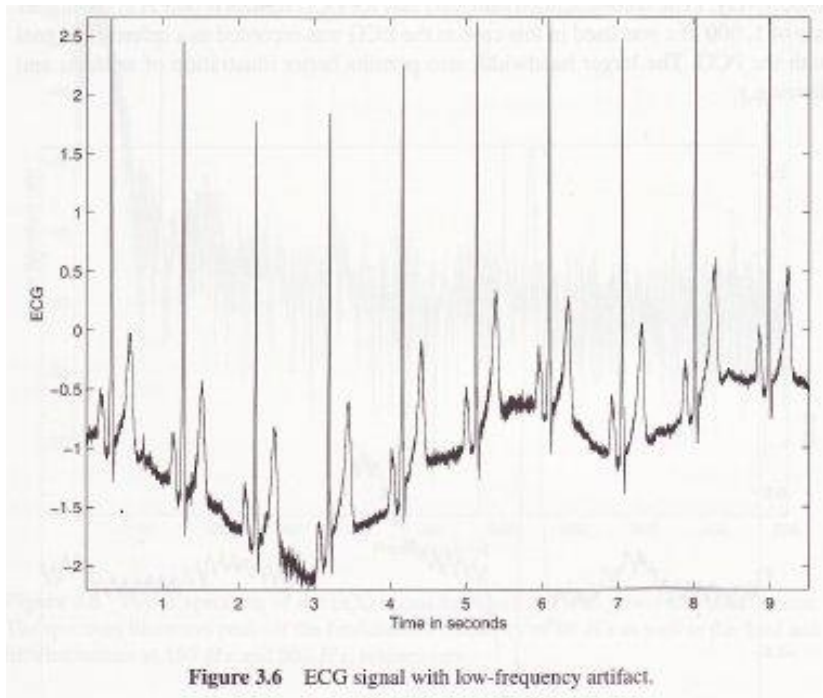
# Trend removal - detrending

- The signal baseline may vary due to, e.g. non-perfect electrode attachment
  - The baseline wandering may disturb analysis of signal properties
  - It is thus favorable to remove the baseline as well if necessary for the application
- High-pass filtering
  - time-domain: difference filter
  - frequency-domain: DFT (discrete Fourier transform)
- Trend removal with other methods
  - Savitzky-Golay filter

# Difference filtering, version 1

First-order difference operator:  
T=sampling interval

$$y(n) = \frac{1}{T}[x(n) - x(n-1)]$$



# Difference filtering, version 2

Modified first-order difference operator:

- T=sampling interval
- Additional pole inserted at zero frequency to steepen the transition band

$$H(z) = \frac{1}{T} \left[ \frac{1 - z^{-1}}{1 - 0.995z^{-1}} \right]$$

$$y(n) = \frac{1}{T} [x(n) - x(n-1)] + 0.995y(n-1)$$

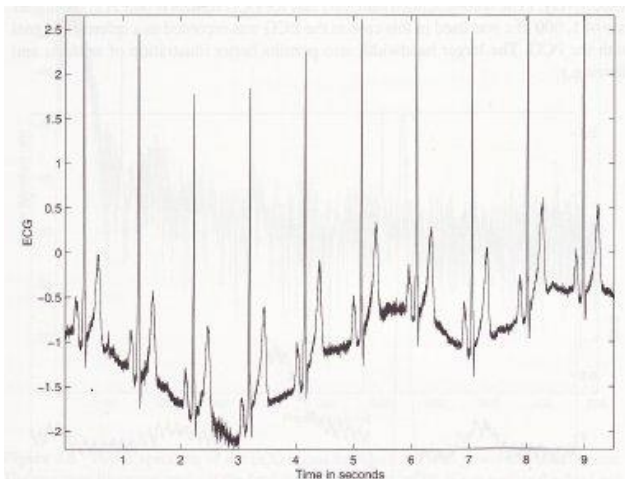


Figure 3.6 ECG signal with low-frequency artifact.

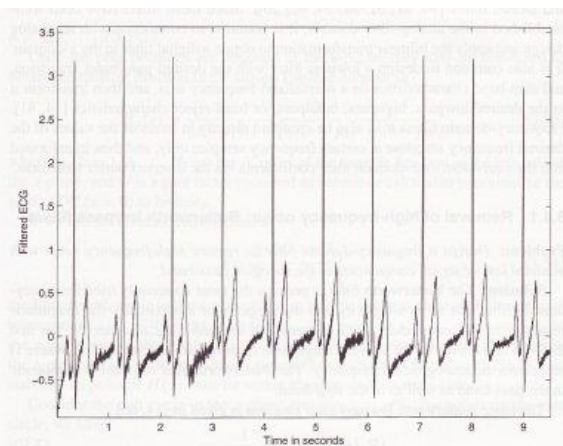


Figure 3.28 Result of processing the ECG signal with low-frequency noise shown in Figure 3.6, using the filter to remove base-line wander as in Equation 3.47. (Compare with the results in Figures 3.24 and 3.25.)

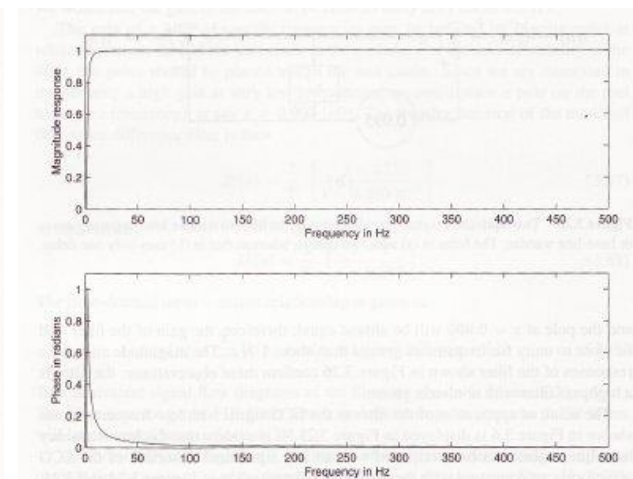
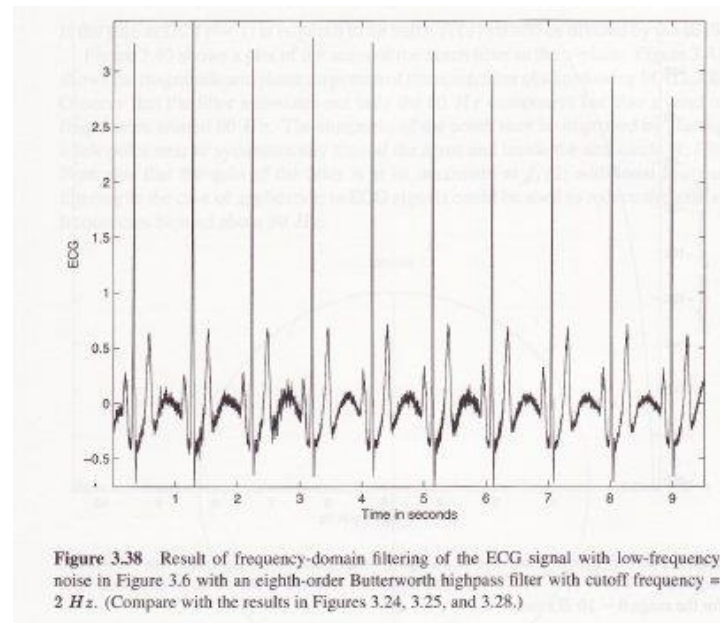
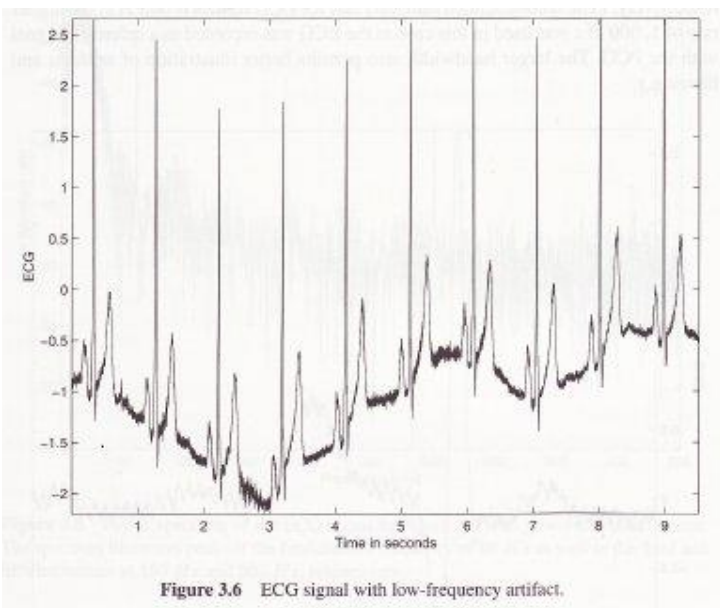
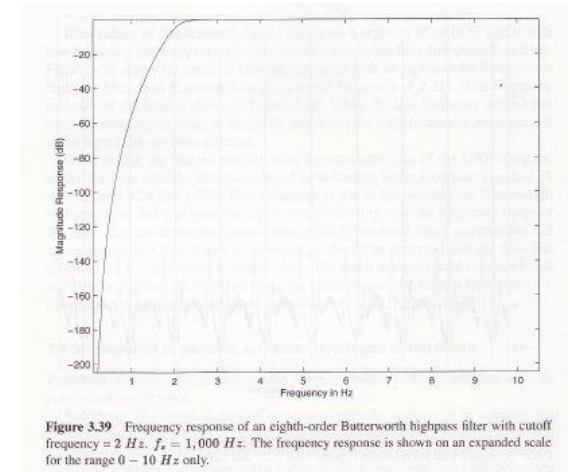


Figure 3.27 Normalized magnitude and phase responses of the filter to remove base-line wander as in Equation 3.47. The magnitude response is shown on a linear scale.



# Detrending: Butterworth highpass filter

Select suitable filter order and cutoff frequency





# Savitzky-Golay filter

- S-G filters are called polynomial or least-squares smoothing filters
- Fits a polynomial of given degree optimally to a signal window
- In a sliding time window (frame), a polynomial curve is fitted to signal, and its middle value in the frame is taken as the smoothed value within the window
- Detrending procedure: subtract the smoothed/filtered signal from the original signal
  - This allows for decomposition of the signal into a trend signal and residual/detail signal
  - The trend component can be interpreted as the useful signal component or the noise component, depending on the application
- Can be implemented as a fast FIR filter

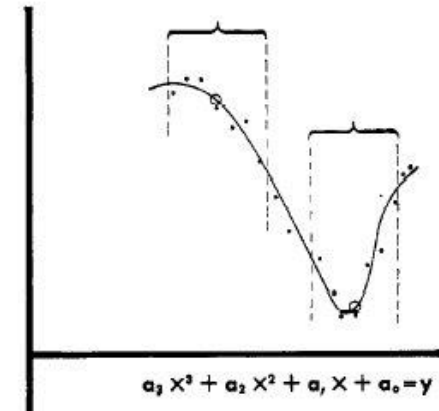
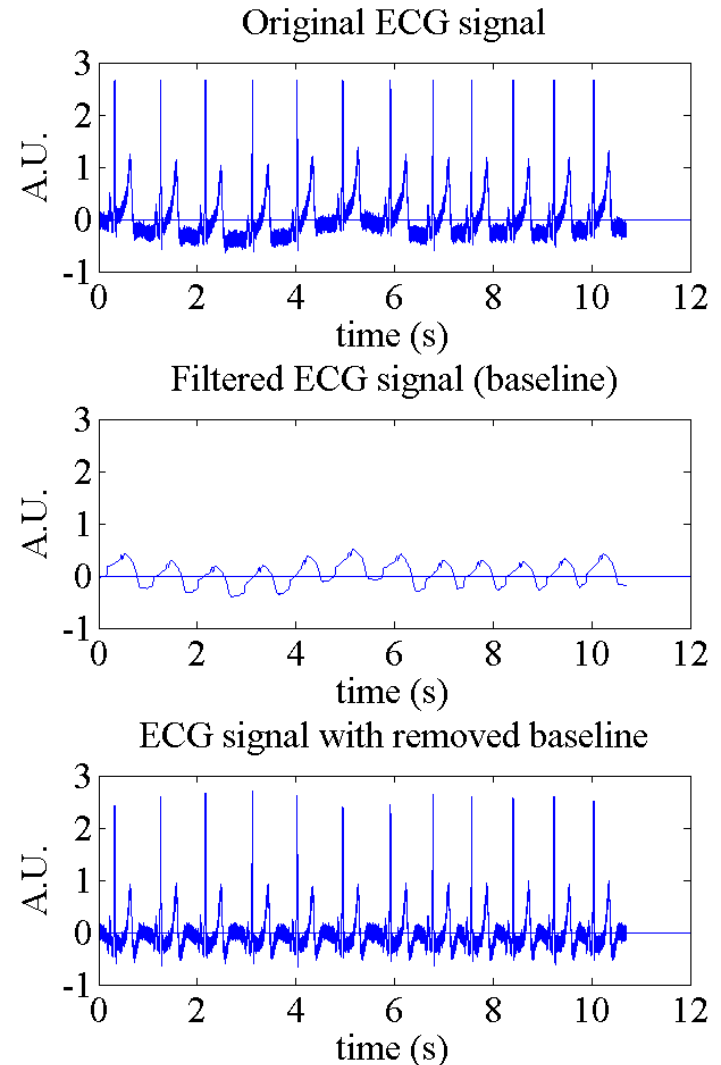


Figure 4. Representation of a 7-point moving polynomial smooth.

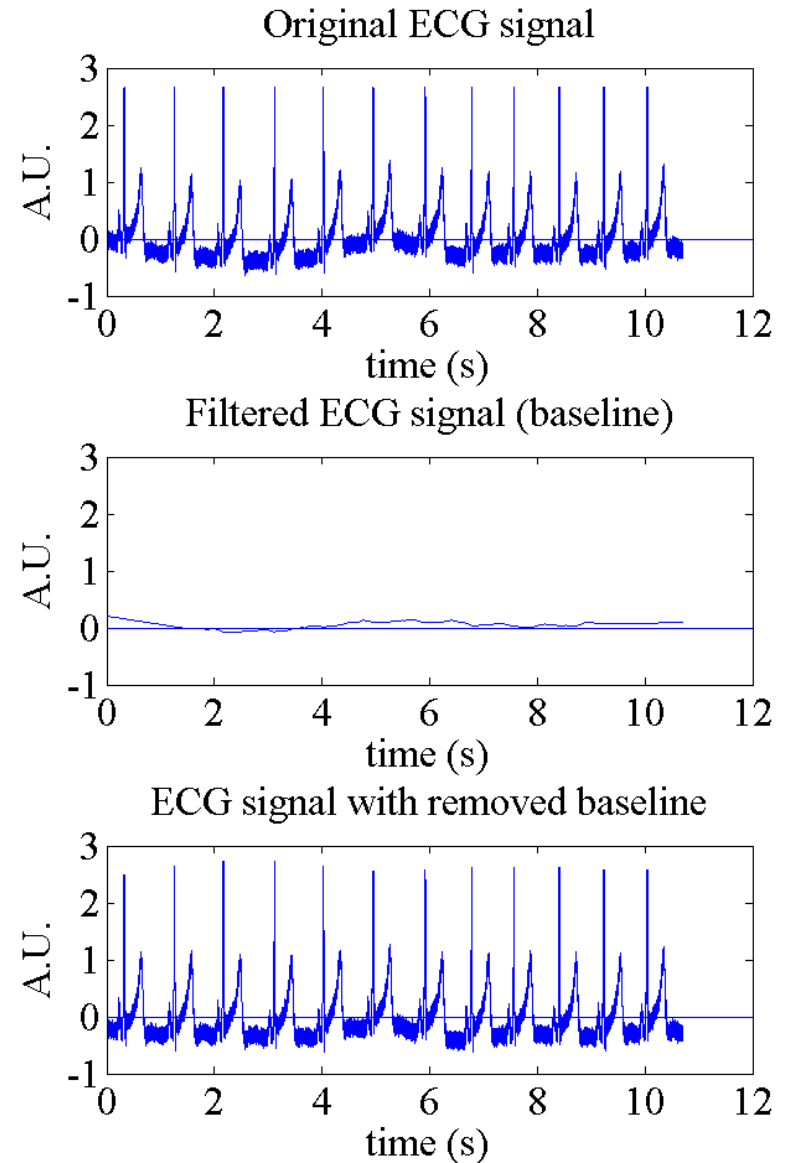
# Savitzky-Golay: detrending example with ECG

- Parameters:
  - Degree of polynomial (usually 1 or 2)
  - Window/frame size
    - depends on signal's timing properties and, thus, sampling frequency
- Parameter selection affects strongly the filtering results:
  - The higher degree the polynomial is, the more the small details are followed and removed in the output signal!
  - Figure on right: too high polynomial was used: the baseline estimate follows ECG shapes too closely and removes important information!



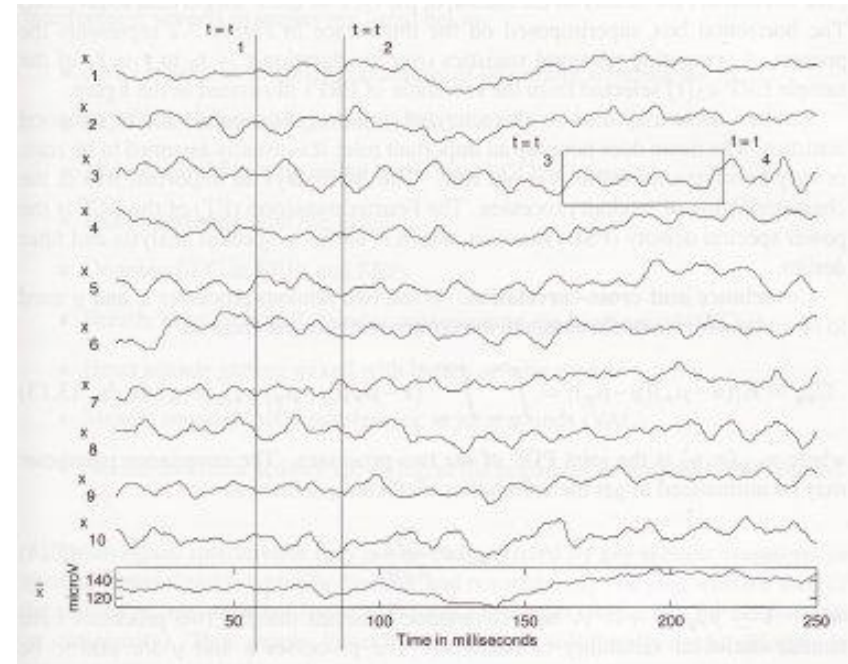
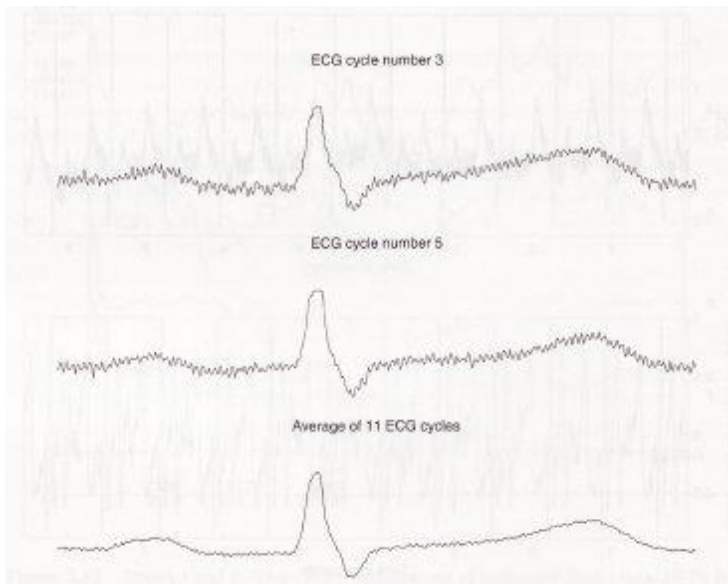
# Savitzky-Golay: detrending example with ECG, cont'd

- Figure on right: more proper polynomial degree (low) was used: the baseline estimate follows the trend better
  - The ECG waveforms are retained better (in the bottom figure)



# Synchronized averaging

- Filter noise by averaging several signals containing the same events
  - Often simple/complex pulses
- Signals must first be time-synchronized



Averaging of flash visual ERP's from EEG

Central limit theorem: the sum of i.i.d. random variables with finite distributions approaches normal distribution. With zero mean variables, the sum approaches zero. Here, the random variables represent noise.

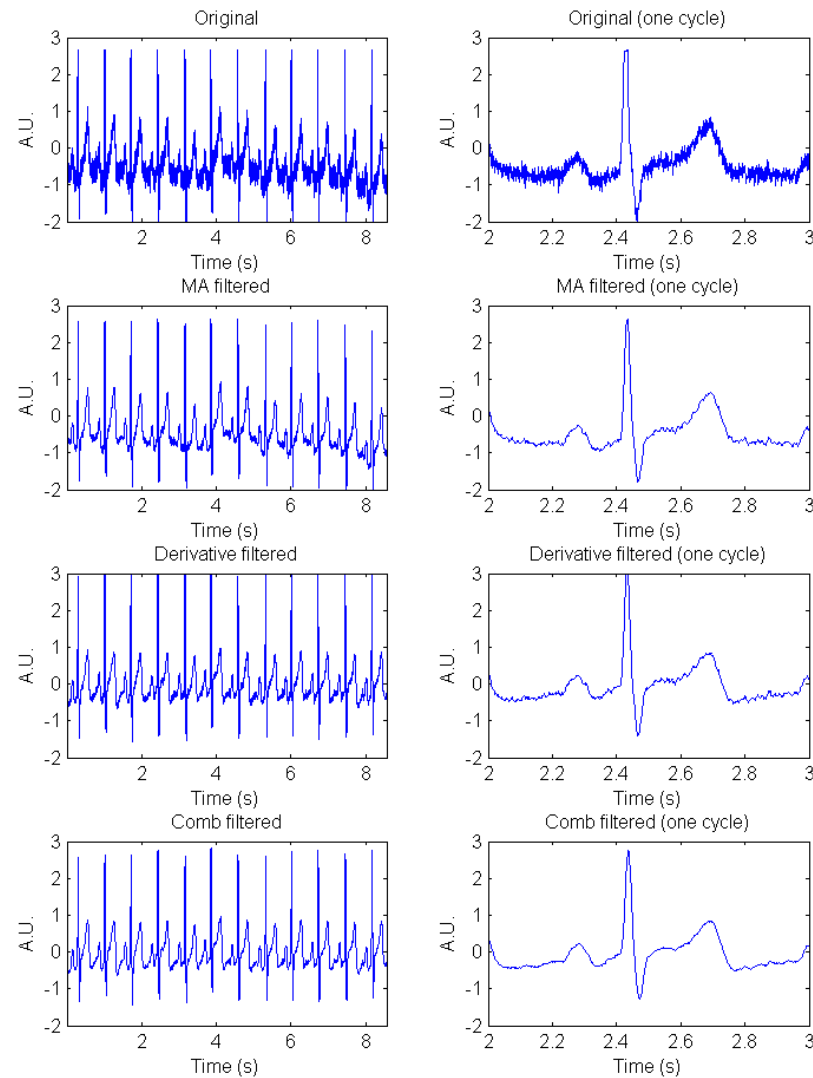
Example case of multi-stage  
filtering

# Filtering many noise types

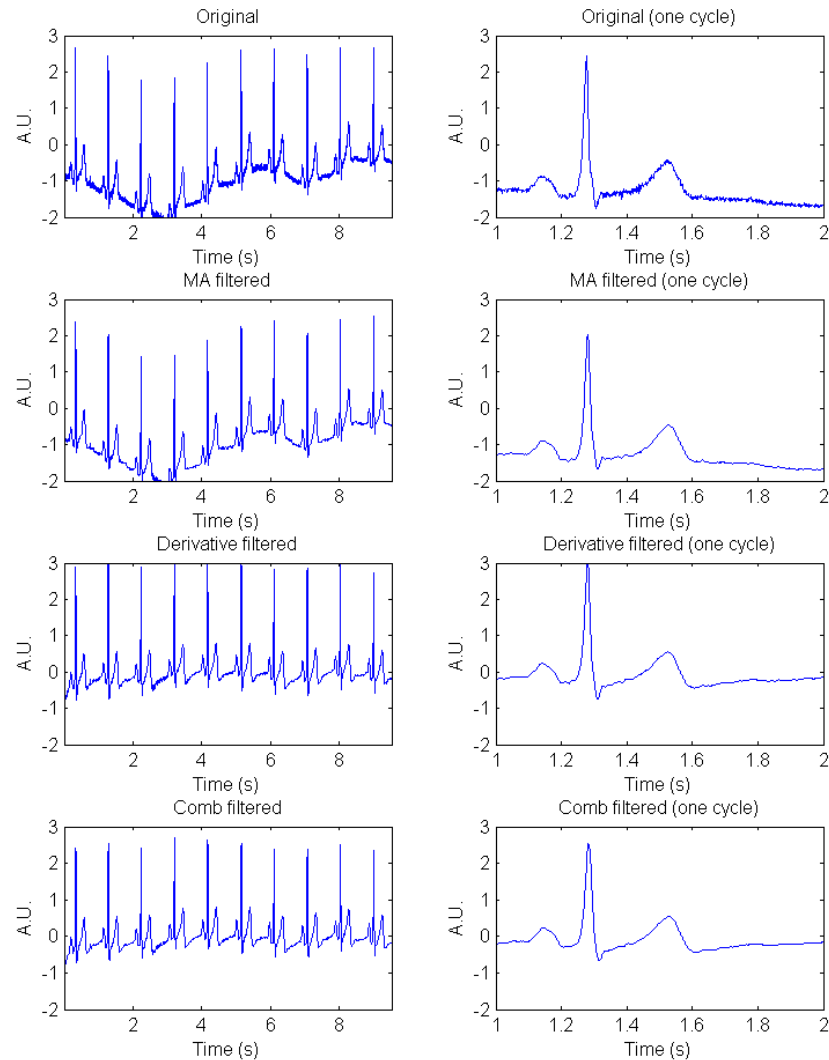
- Often the signal contains different kinds of noise
- A pipeline must be designed so that each stage removes one type of noise
- Filter stages can sometimes be combined into one stage
  - E.g.: LP + HP  $\rightarrow$  BP
- An example filter pipeline:



# Filtering many noise types: example result 1



# Filtering many noise types: example result 2





# Selected references

Course book: Chapter 3

## Journal article

- Savitzky A, Golay MJE (1964) Smoothing and Differentiation of Data by Simplified Least Squares Procedures. Anal Chem 36(8):1627–1639.

## Books on signal processing basics

- Ifeachor EC, Jervis BW. Digital Signal Processing: A Practical Approach. Addison-Wesley, reprint 1996, pp. 279-287, 375-383, 550-551, 561-563, 697-706.
- Orfanidis, SJ. Introduction to Signal Processing. Prentice-Hall, Englewood Cliffs, NJ, 1996, pp. 434-441.