



**KAPITAŁ LUDZKI**  
NARODOWA STRATEGIA SPÓJNOŚCI

**UNIA EUROPEJSKA**  
EUROPEJSKI  
FUNDUSZ SPOŁECZNY



## **„SIGNAL PROCESSING”**

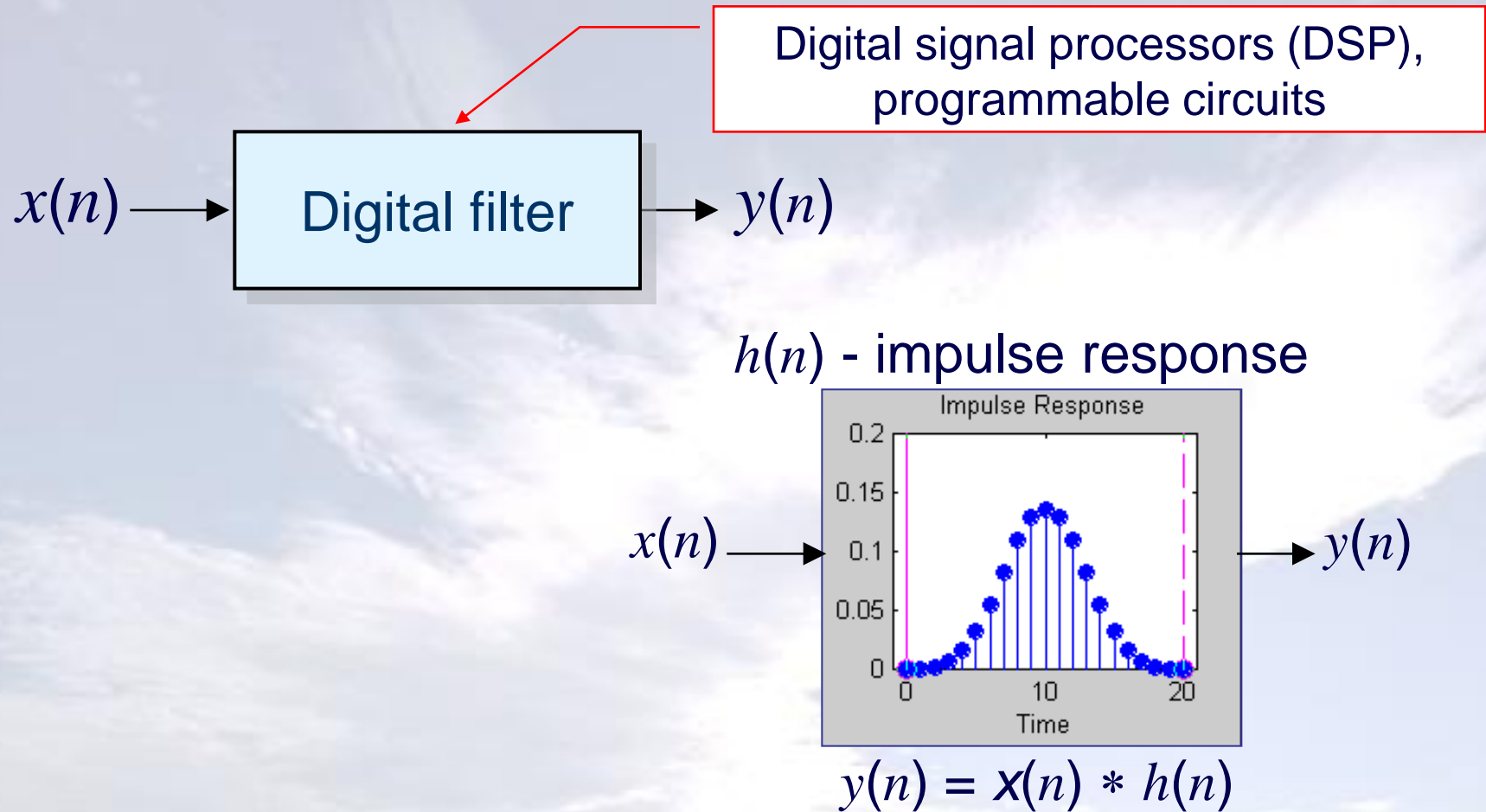
**Prezentacja multimedialna współfinansowana przez  
Unię Europejską w ramach  
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*„Innowacyjna dydaktyka bez ograniczeń - zintegrowany  
rozwój Politechniki Łódzkiej - zarządzanie Uczelnią,  
nowoczesna oferta edukacyjna i wzmacniania zdolności  
do zatrudniania osób niepełnosprawnych”***



Politechnika Łódzka

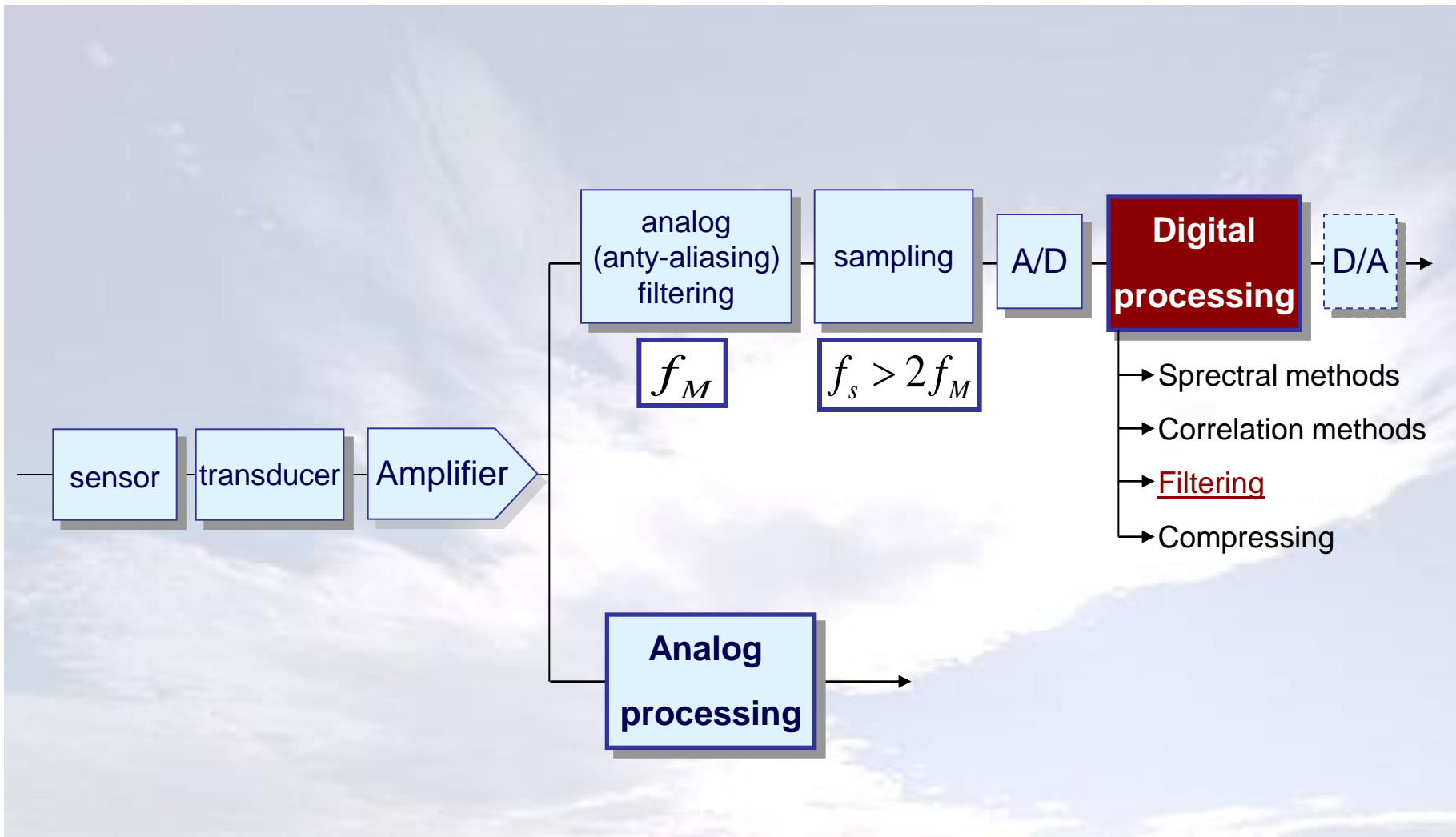
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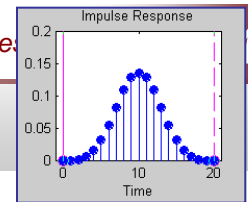
# Digital filters





# Signal processing





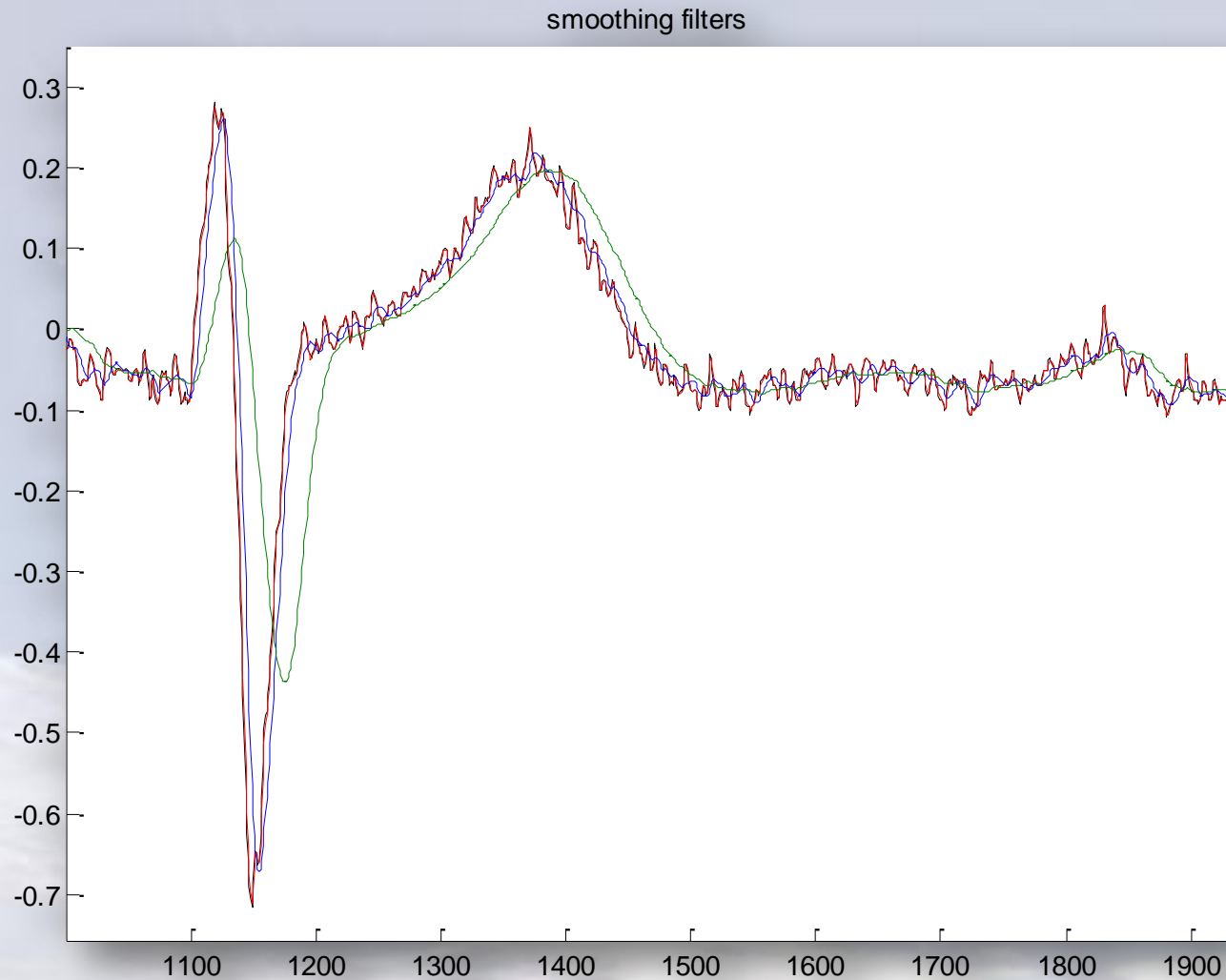
## Why to filter the signal?

- to reduce noise (eg. from the mains network)
- to change the spectral characteristic of the signal (*preemphasis, deemphasis*)
- to filter out the components of interest from the given signal (detection)



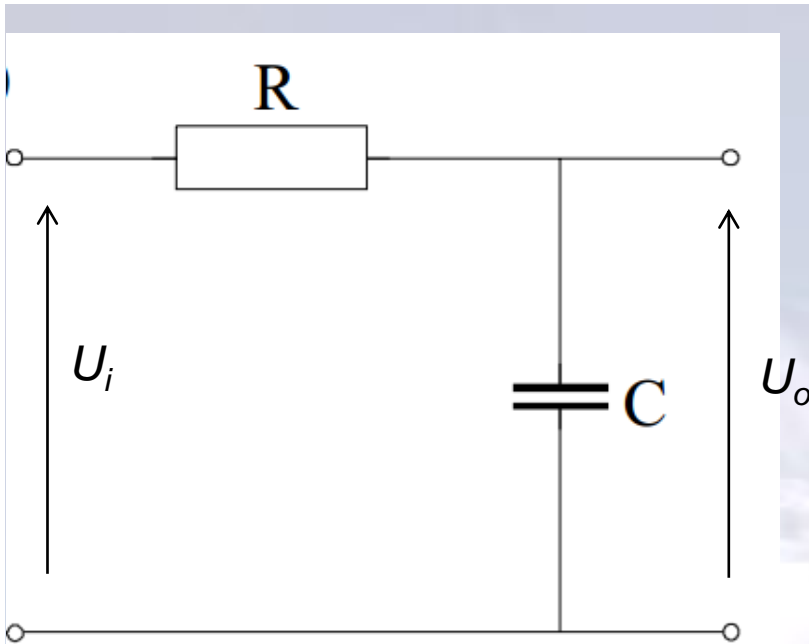


# Signal smoothing by filtering



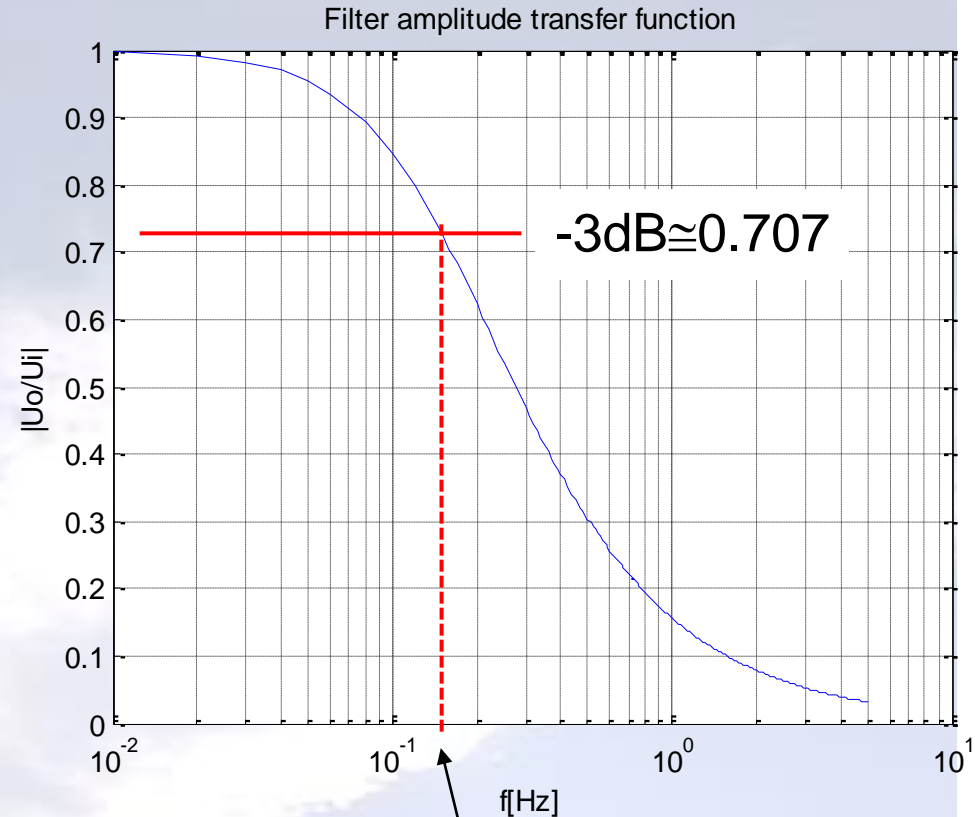


# An analogue low-pass filter example



Amplitude transfer function:

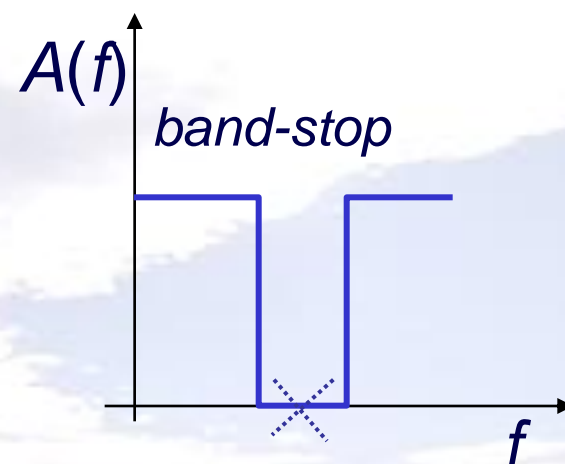
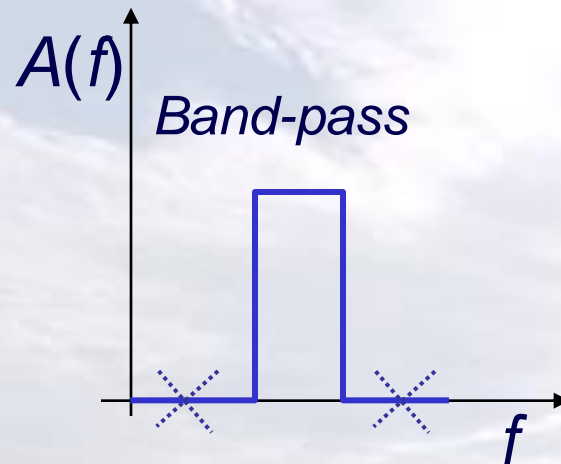
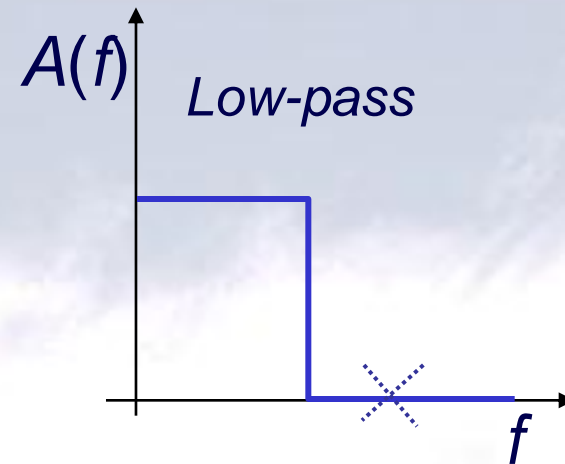
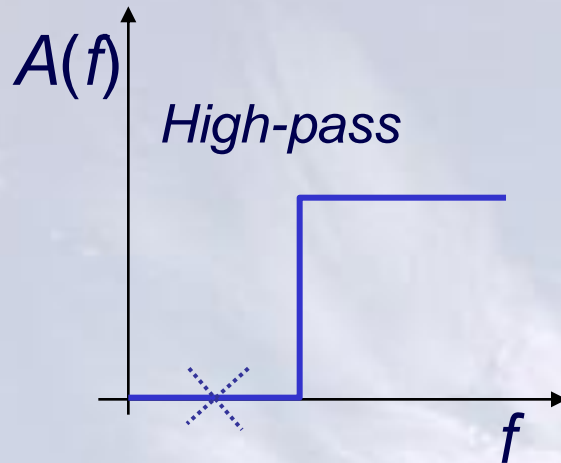
$$\left| \frac{U_o}{U_i} \right| = \frac{1}{\sqrt{1 + (\omega RC)^2}}$$



$$f_c = ?$$



# Frequency characteristics – ideal filters

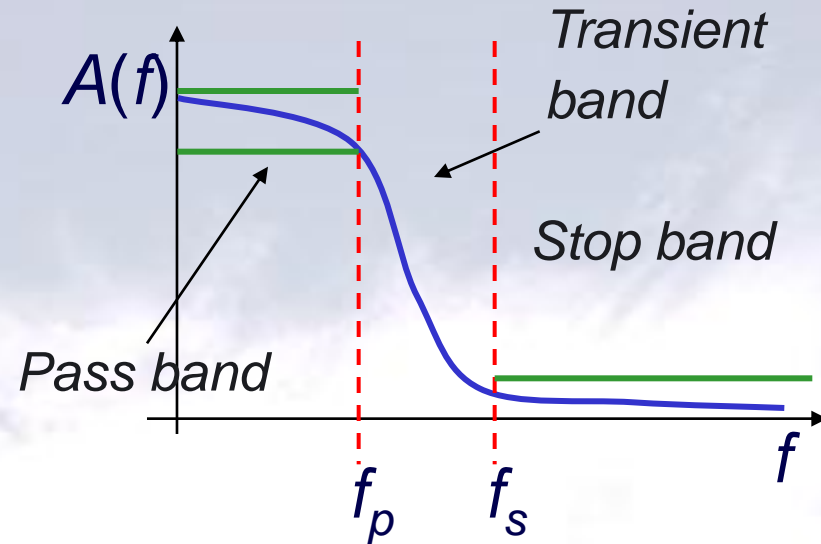




# Frequency characteristics

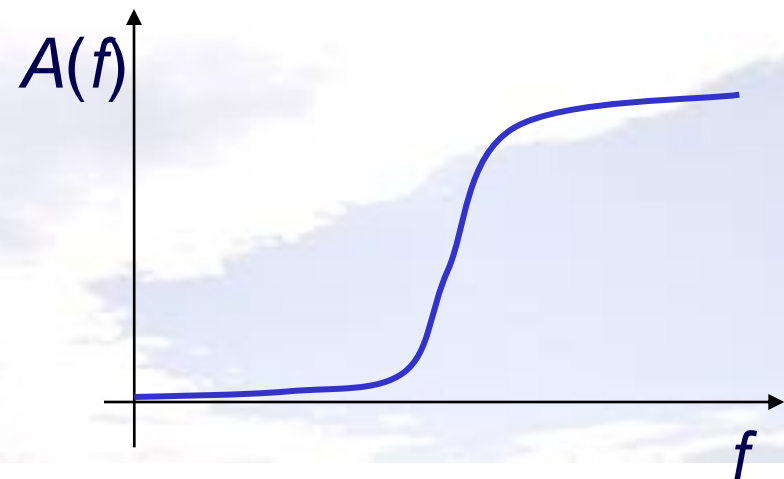
## Low-pass filter

(eg. *antialiasing filter*,  
*noise reduction*)



## High-pass filter

(eg. *preemphasis*)



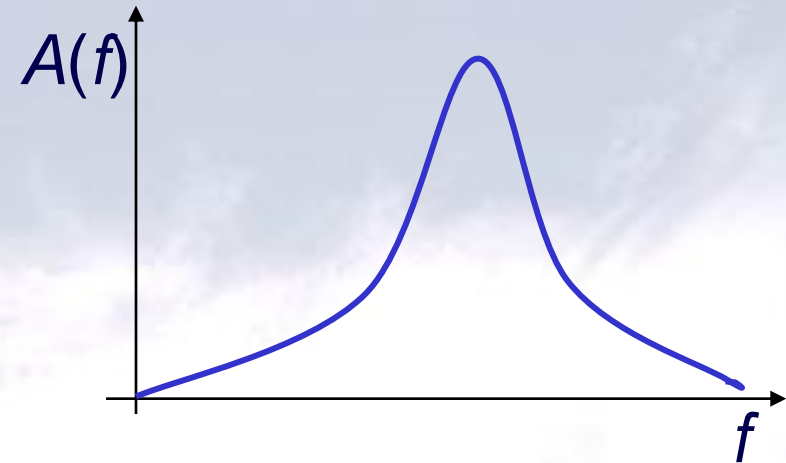




# Frequency characteristics

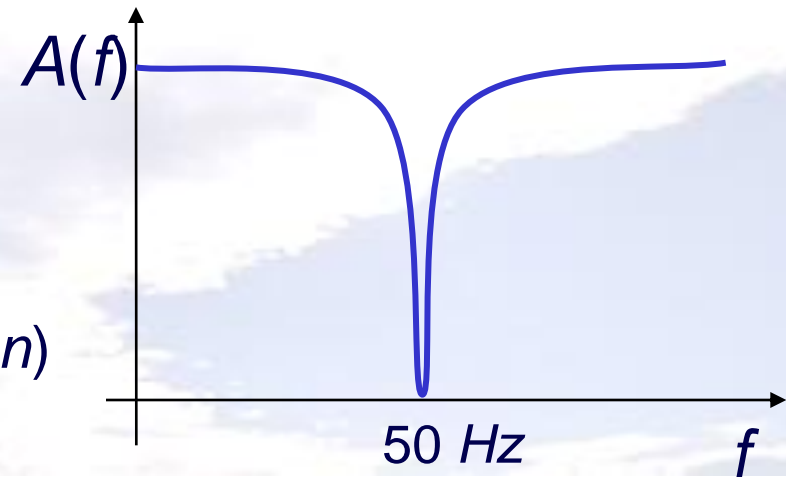
## Band-pass filter

(eg. *feature detection*)

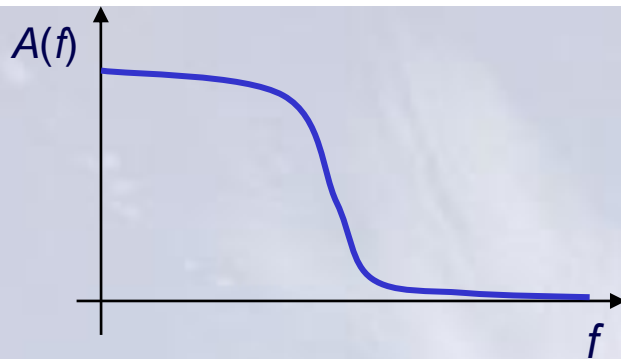


## Band-stop filter

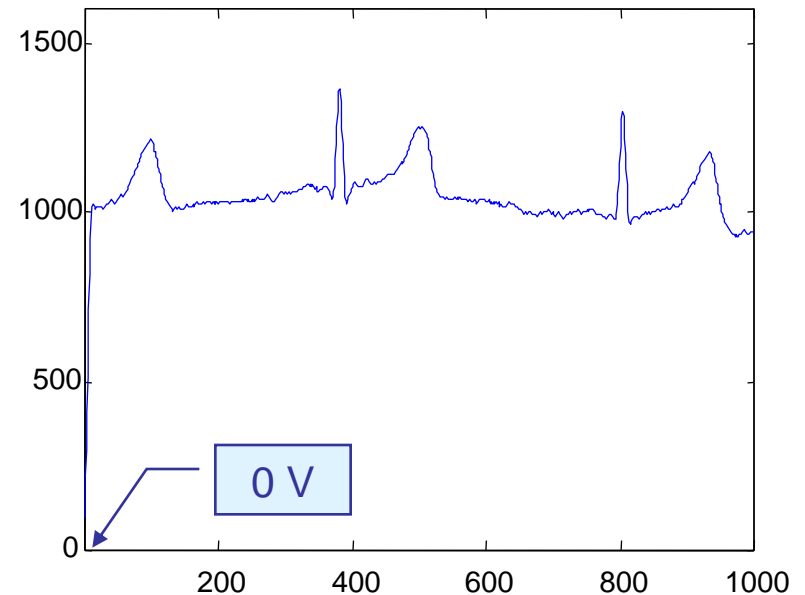
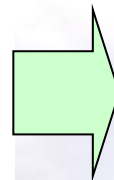
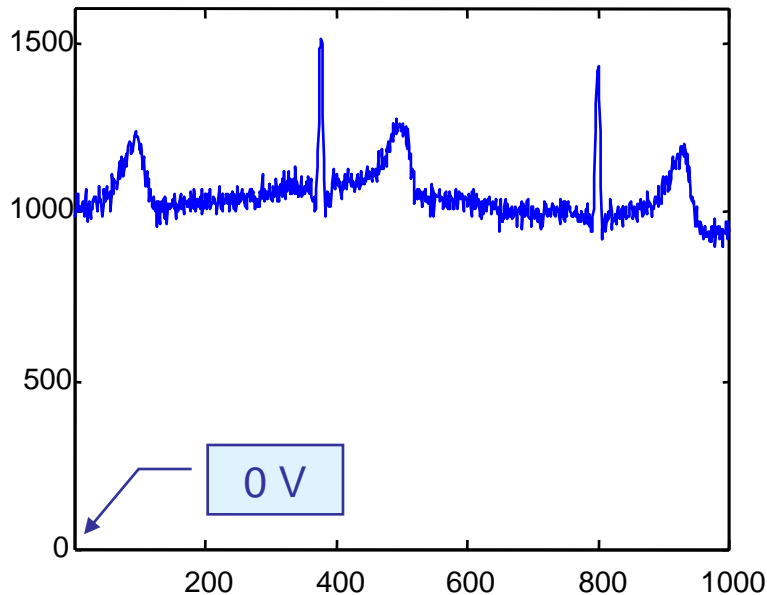
(eg. *mains interference reduction*)



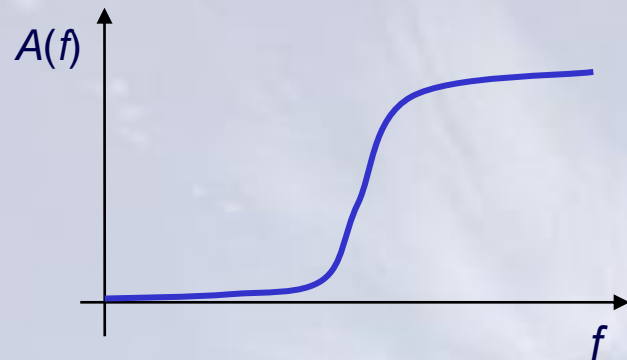
# Frequency characteristics - examples



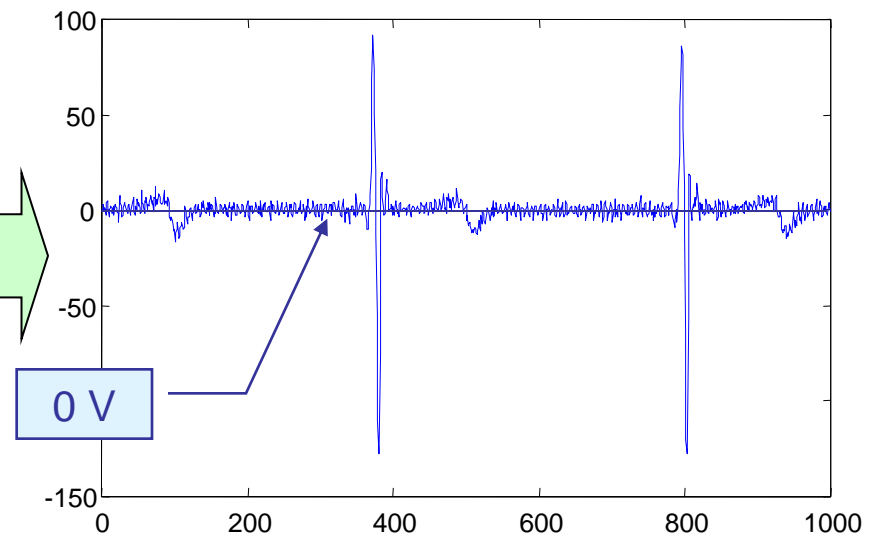
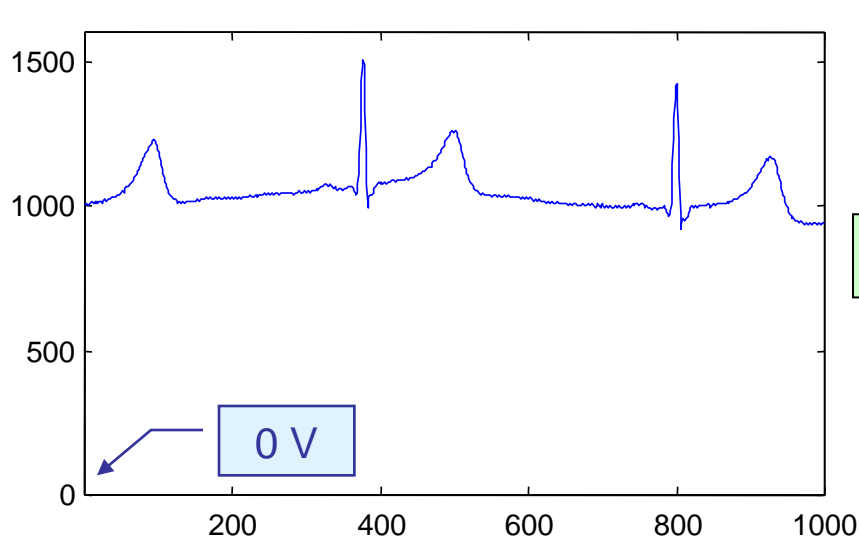
Low-pass filter  
(noise reduction)



# Frequency characteristics - examples

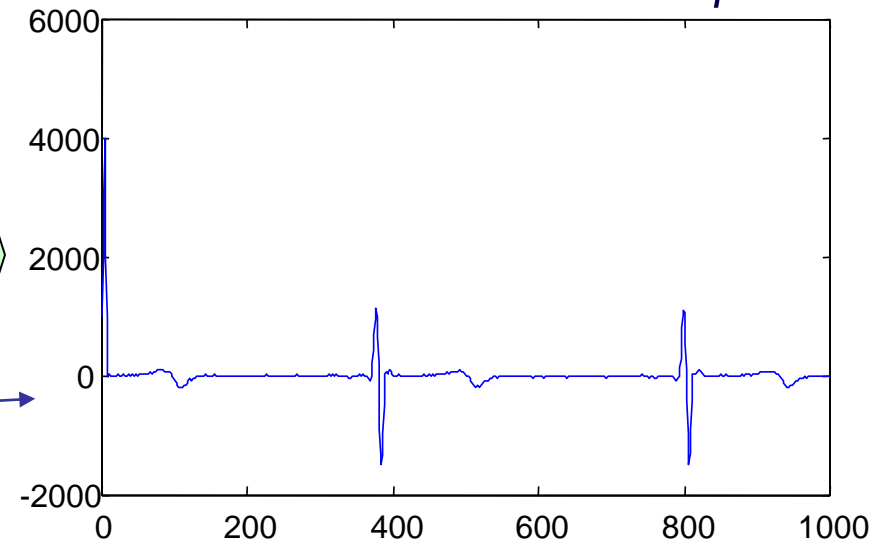
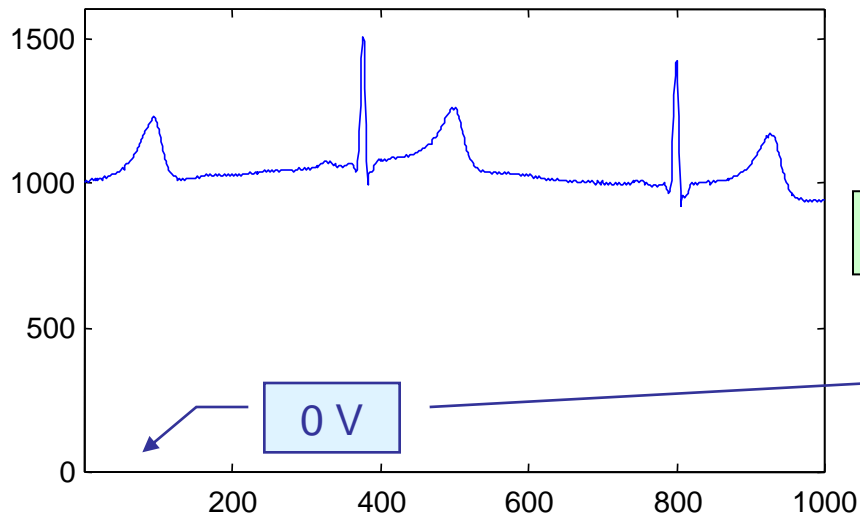
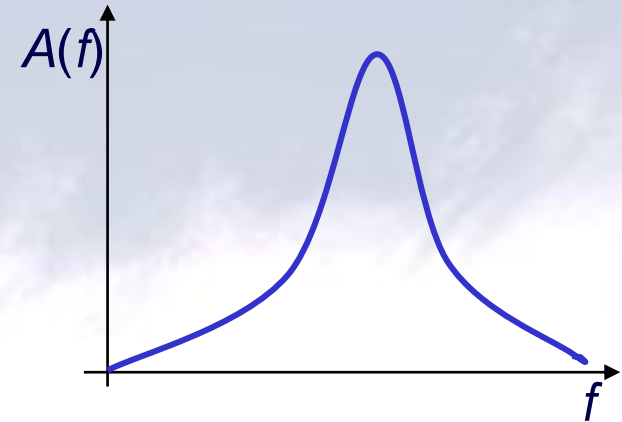


High-pass filter  
(eg. *mean subtraction*)

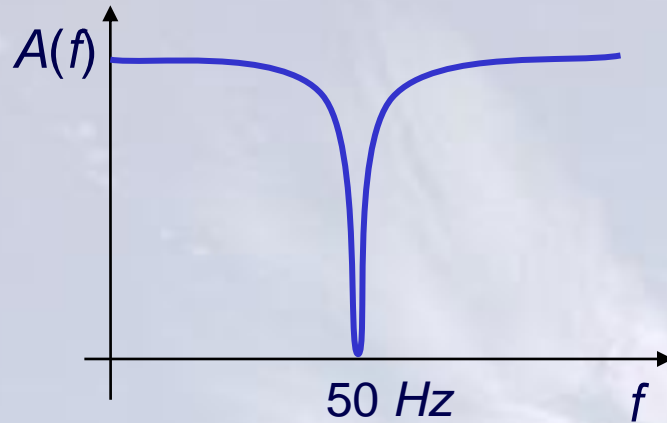


# Frequency characteristics - examples

## Narrow band-pass filter (eg. *feature detection*)

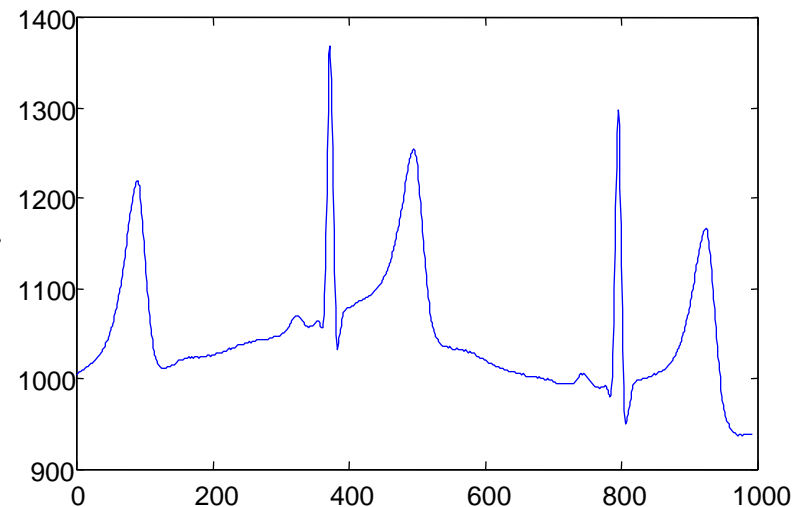
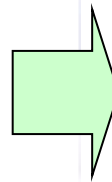
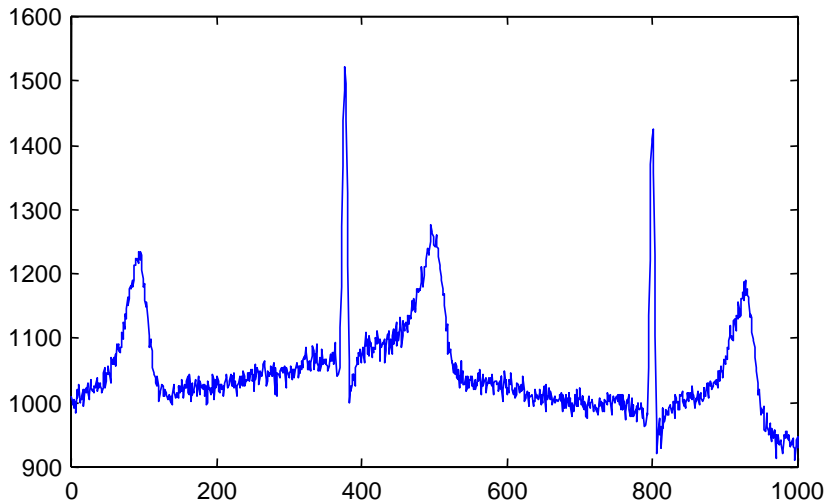


# Frequency characteristics - examples

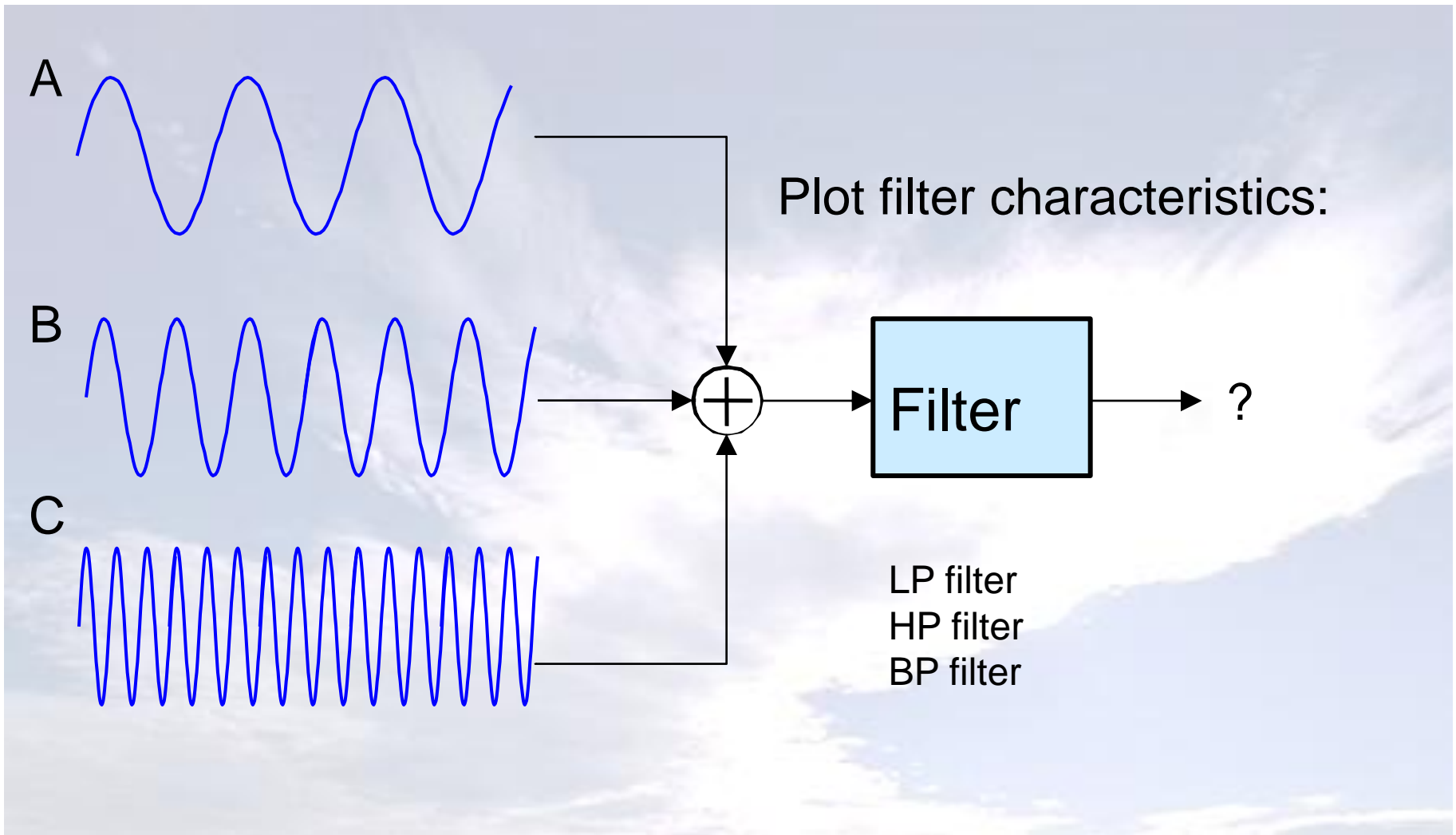


## Band-stop filter

(eg. *Reduction of noise of precise frequency*)



# Filter types



# Application of filters to ECG processing

- **lowpass** (radio noise reduction, reduction of skeletal muscles activity noise)
- **highpass** (elimination of izoelectric line migration,  $f_g=0.5$  Hz, see. 'ecg\_mit.mat')
- **bandpass** (ECG feature detection, eg. P, T, QRS waves)
- **bandstop** (mains network interference reduction ,  $f=50$  Hz)

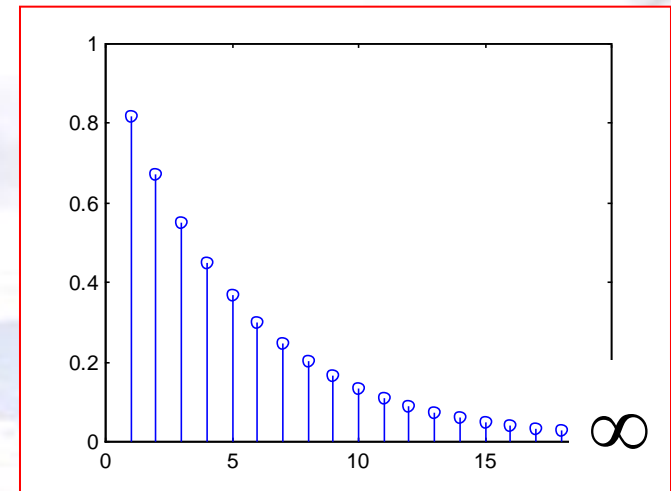
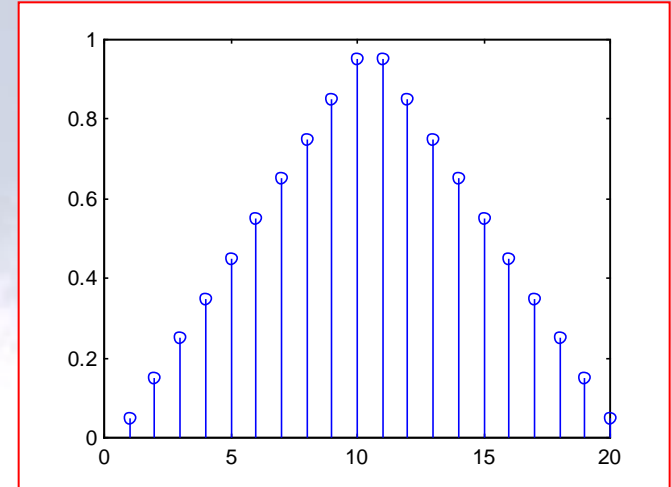
# Digital filters – FIR i IIR

Another division:

**Finite Impulse Response (FIR)**

**Infinite Impulse Response (IIR)**

*co called **recursive filters***

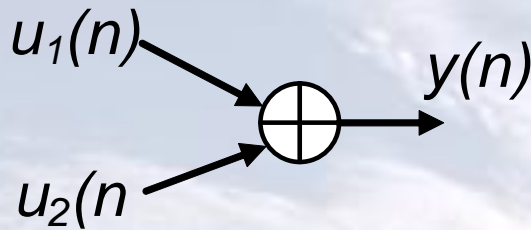




# Digital filters – FIR i IIR

The following elementary functional operators are used in the design of a digital filter :

Summation



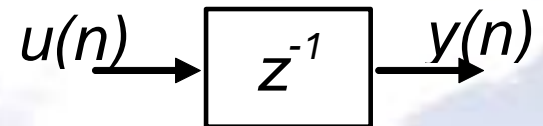
$$y(n) = u_1(n) + u_2(n)$$

Multiplication  
by a constant



$$y(n) = cu(n)$$

Unit delay

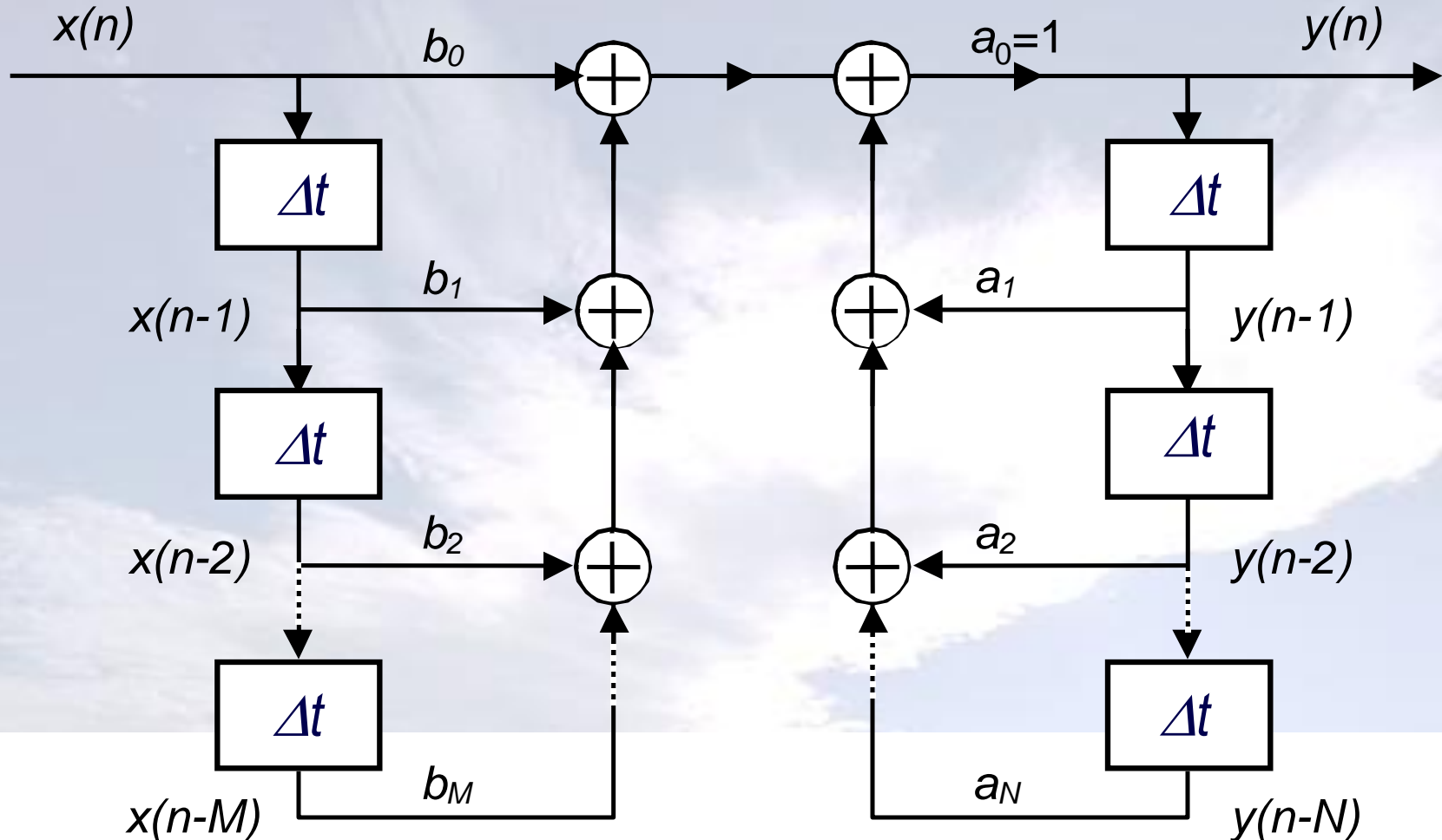


$$y(n) = u(n-1)$$

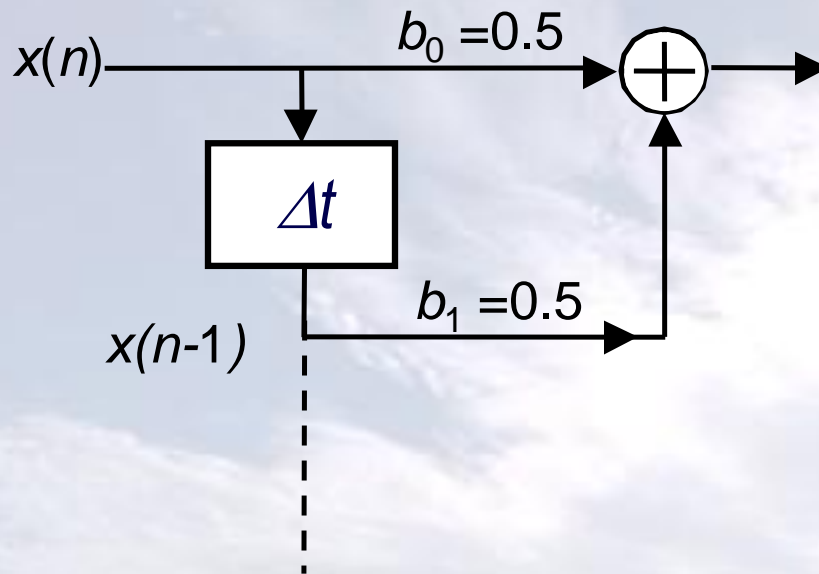
# The idea of digital filtering

*Moving average coefficients*

*Autoregression coefficients*



# Finite impulse response (FIR) filter - example



LP:

$$y(n) = 0.5x(n) + 0.5x(n-1)$$

HP:

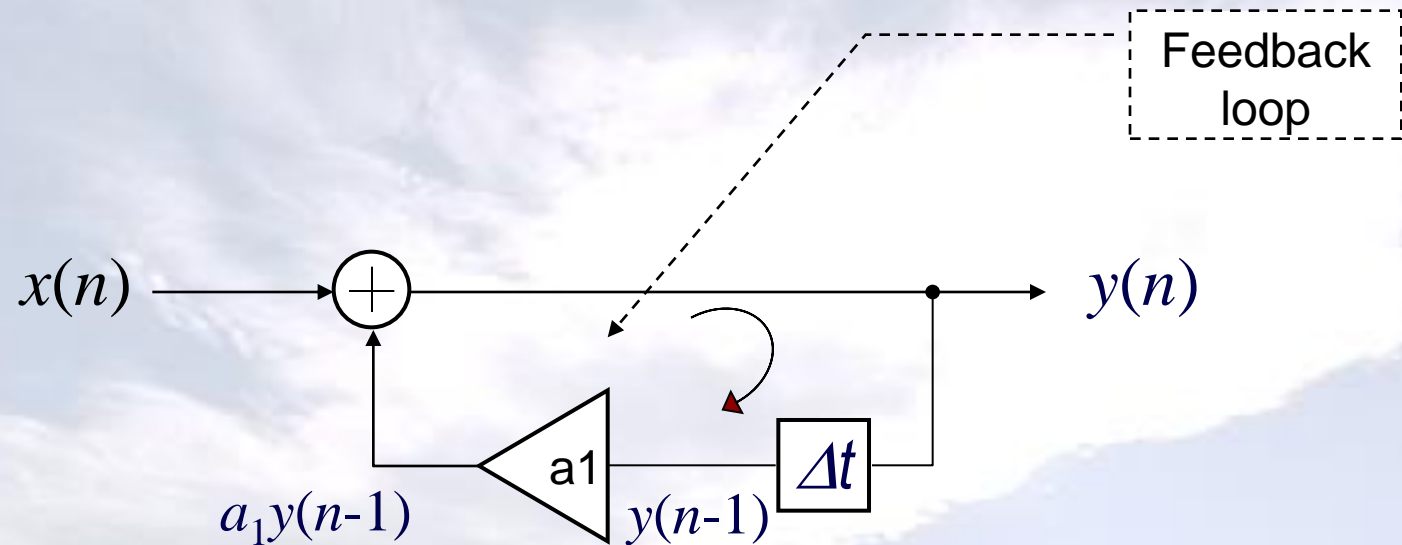
$$y(n) = 0.5x(n) - 0.5x(n-1)$$

BP: ?



# Infinite response filter (IIR) - example

$$y(n) = a_1 y(n-1) + x(n)$$



# Difference equation of a digital filter

$$\sum_{k=0}^N a(k)y(n-k) = \sum_{k=0}^M b(k)x(n-k)$$

which is equivalent to a difference equation:

$$a[0]=1$$

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

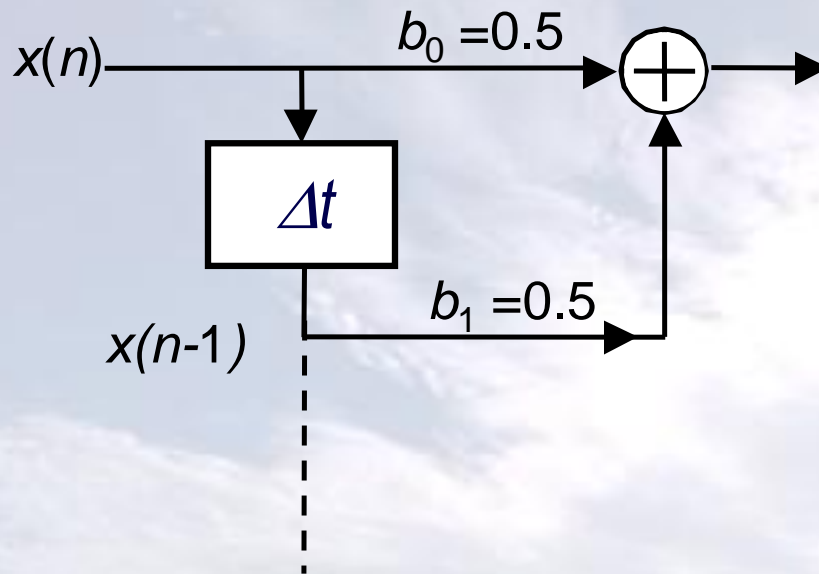
## Difference equation of a digital filter

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

If all  $a(k)$  coefficients are equal zero, then the difference equation defines a FIR filter and an IIR filter otherwise.



# Finite response filter (FIR) - example



LP:

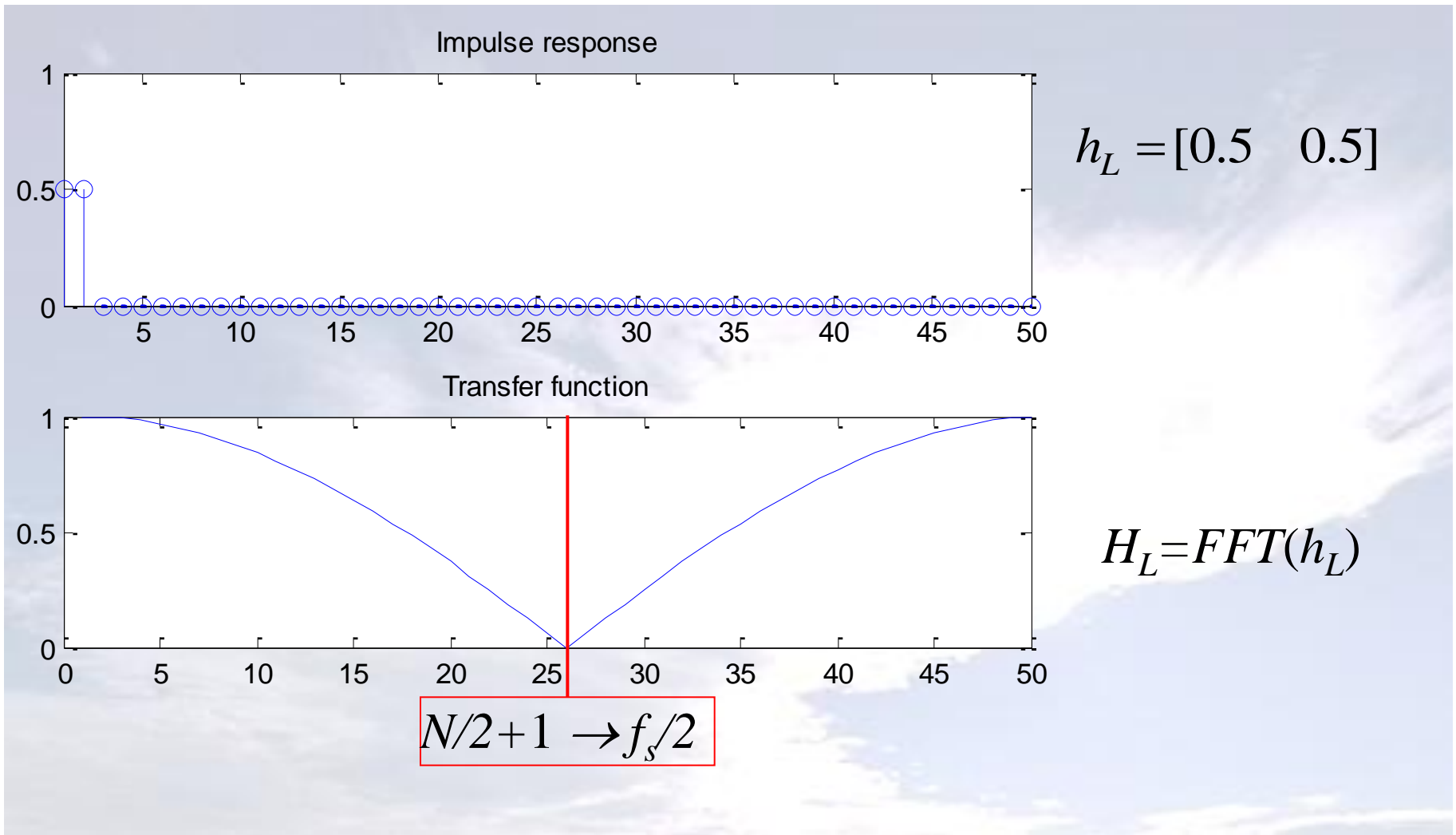
$$y(n) = \overset{b_0}{0.5}x(n) + \overset{b_1}{0.5}x(n-1)$$

HP:

$$y(n) = \overset{b_0}{0.5}x(n) - \overset{b_1}{0.5}x(n-1)$$

BP: ?

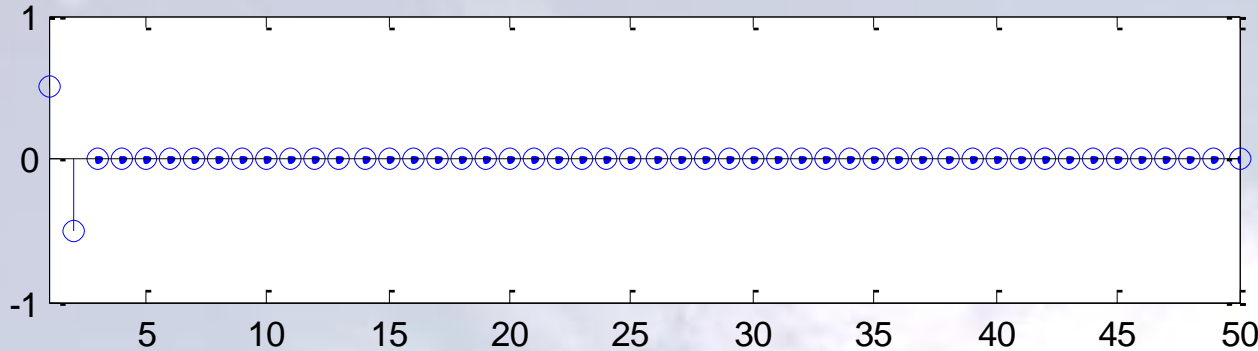
# Finite response filter (FIR) – low pass filter





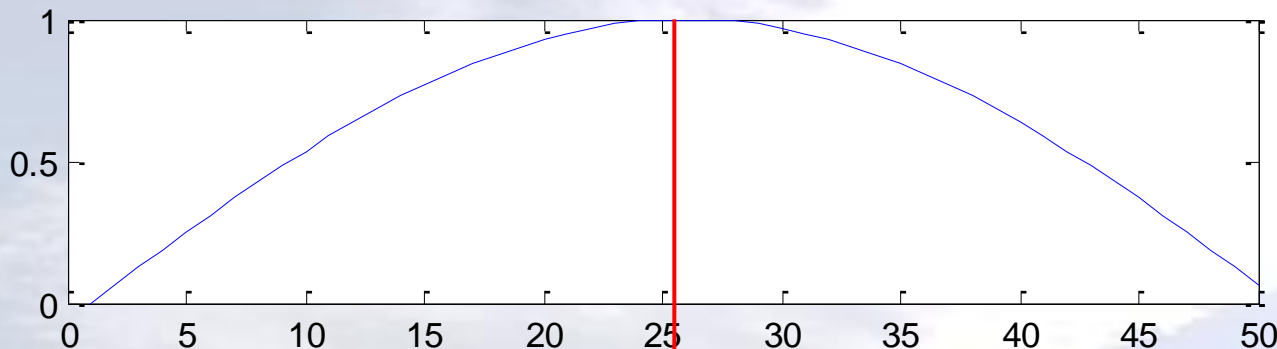
# Finite response filter (FIR) – high pass filter

Impulse response



$$h_H = [0.5 \quad -0.5]$$

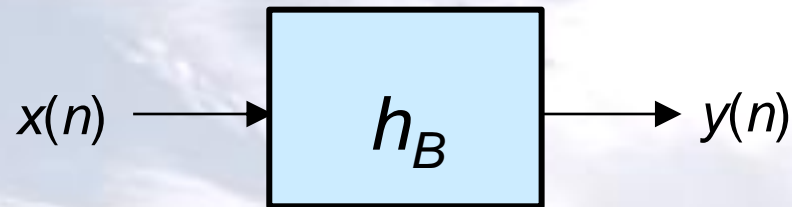
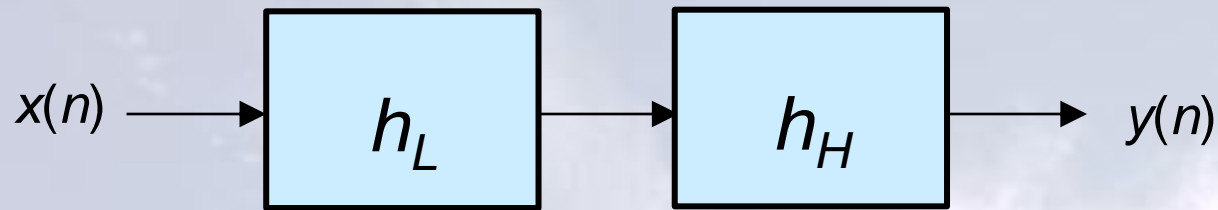
Transfer function



$$H_H = FFT(h_H)$$

$$N/2 + 1 \rightarrow f_s/2$$

# Filters in a cascade connection



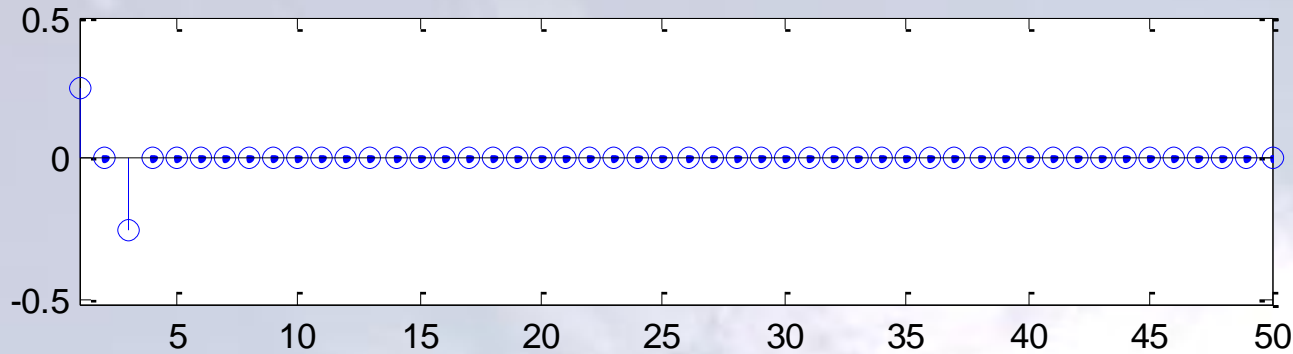
$$h_B = h_L * h_H$$

$$h_B = h_L * h_H \rightarrow H_B(\omega) = H_L(\omega) \cdot H_H(\omega)$$



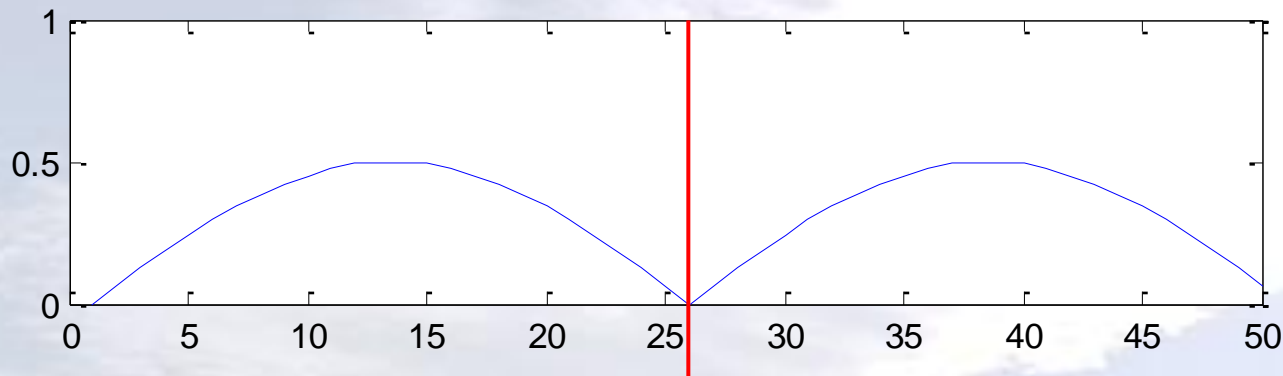
# Finite response filter (FIR) – band pass filter

Impulse response



$$h_B = [0.25 \quad 0 \quad -0.25]$$

Transfer function



$$H_B = FFT(h_B)$$

$$N/2 + 1 \rightarrow f_s/2$$



## Simple FIR filter example

Moving average filter equation is given:

$$y(n) = \frac{1}{5} [x(n-4) + x(n-3) + x(n-2) + x(n-1) + x(n)]$$

Determine the vectors of coefficients  $a$  and  $b$  for this filter .

Solution (simple low pass FIR filter):

$$a=[1];$$

$$b=[0.2 \ 0.2 \ 0.2 \ 0.2 \ 0.2];$$



## Simple FIR filter example

What is the impulse response of this filter:

$$y(n) = \frac{1}{5} [x(n-4) + x(n-3) + x(n-2) + x(n-1) + x(n)]$$

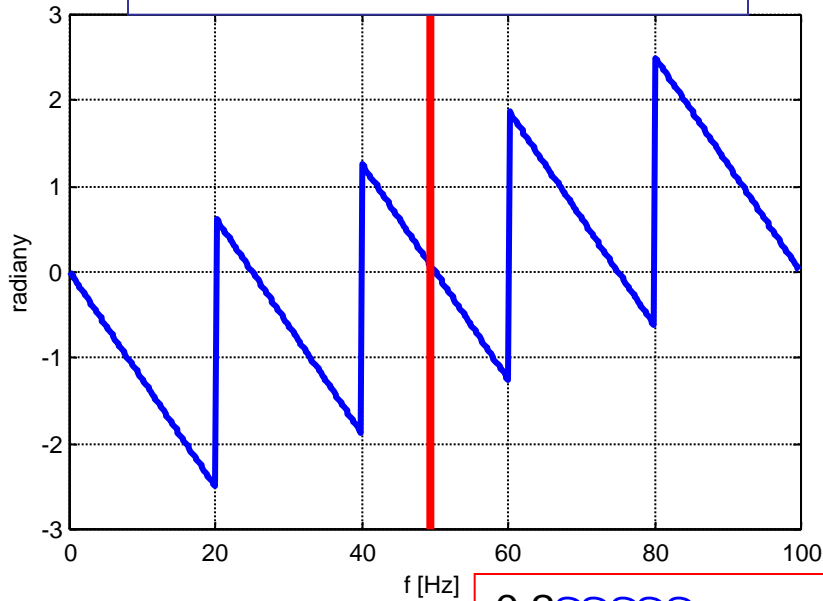
Answer:

$$h = [0.2 \ 0.2 \ 0.2 \ 0.2 \ 0.2];$$

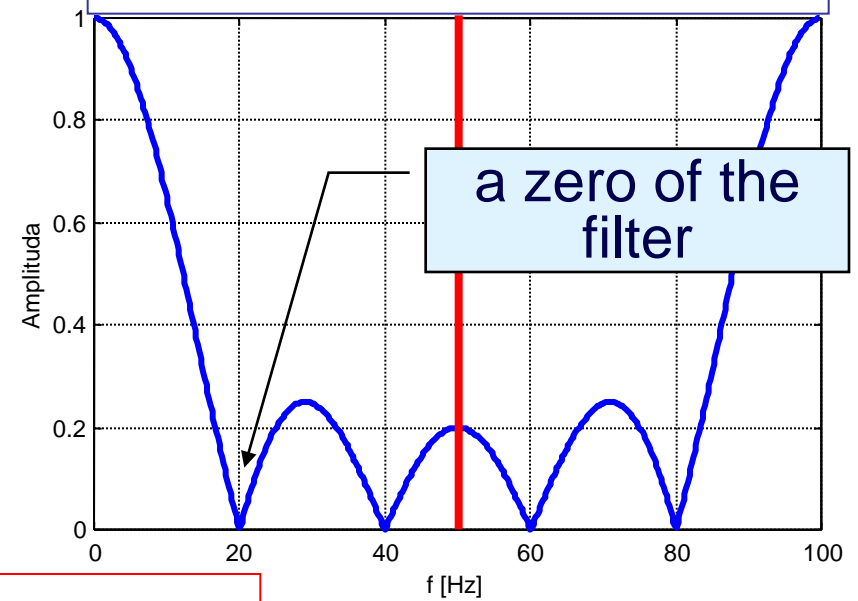
**Important conclusion:** For FIR filters the filter coefficients are equal to filter's impulse response

# Simple FIR filter example

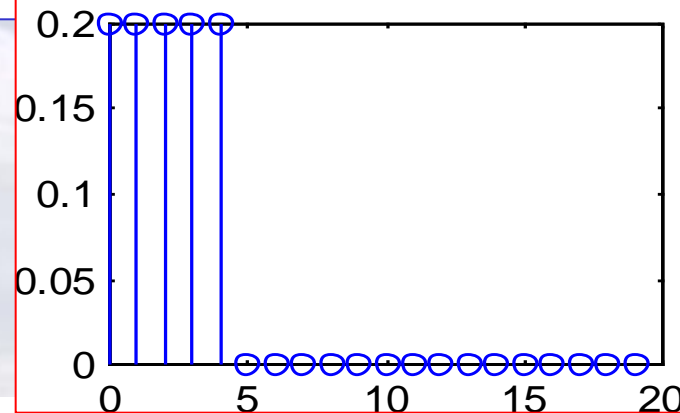
Phase characteristic



Amplitude characteristic



Impuse response of the filter





# Simple FIR filter example

## Exercise (cont.):

Observe the amplitudes and phase shifts of the output characteristics of the filter for the following inputs:

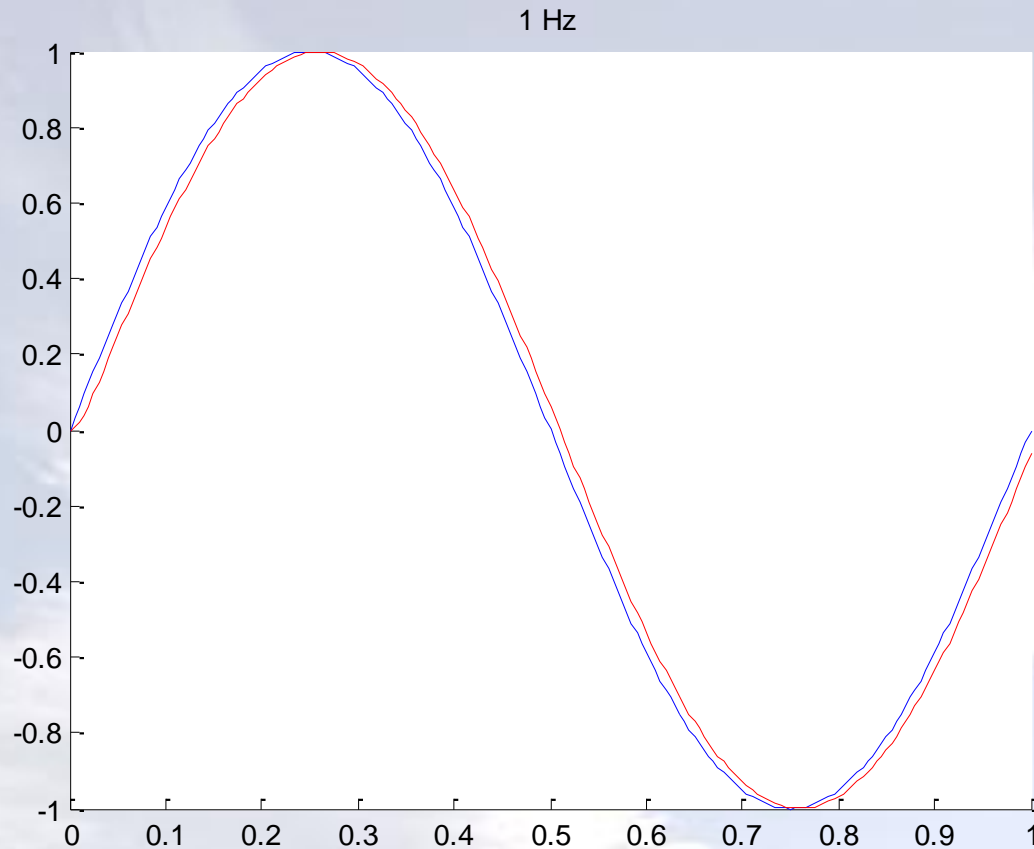
- sinusoid with  $f=1$  Hz
- sinusoid with  $f=5$  Hz
- sinusoid with  $f=20$  Hz

Confirm that filtering (convolution) in time domain is equivalent to multiplication of spectrum of the impulse response of the filter with the spectrum of the signal.





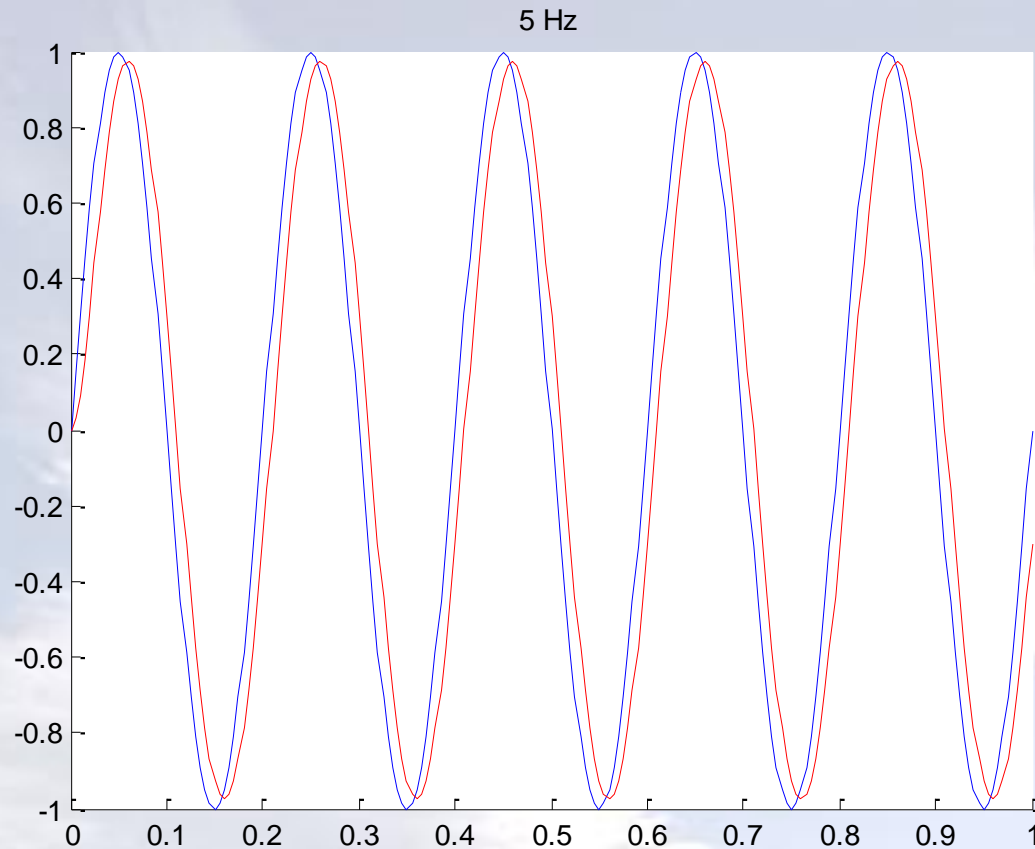
# Simple FIR filter example





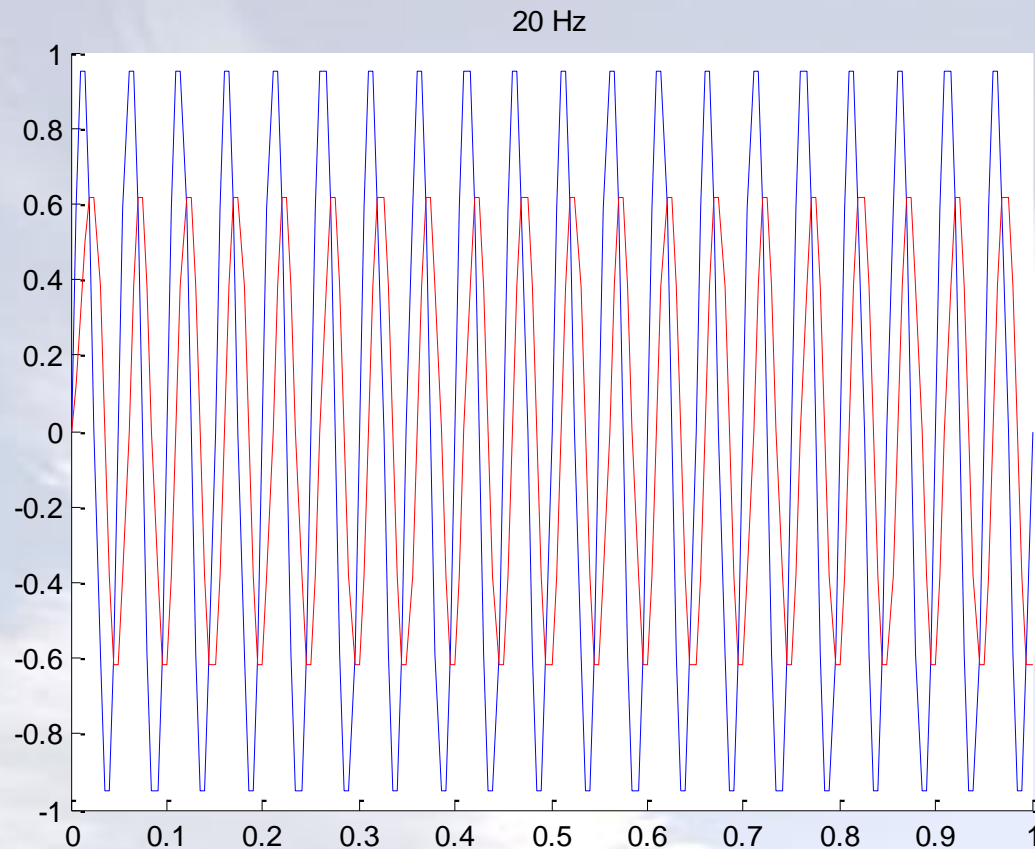


# Simple FIR filter example



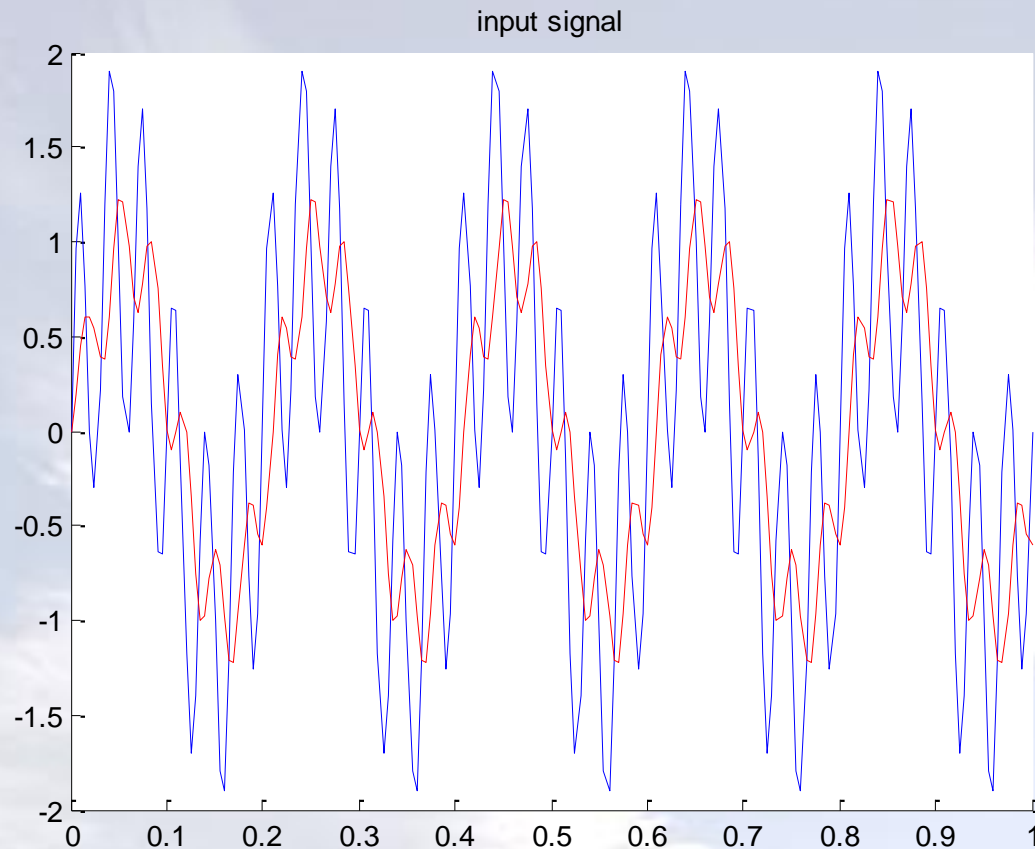


# Simple FIR filter example





# Simple FIR filter example



## Linear phase condition for FIR filters

FIR filters may be designed so that their phase characteristic is linear. For this purpose the 'symmetry' condition for the filter coefficients is to be fulfilled.

$$h(M - k) = h(k) \quad \text{for } k = 0, 1, \dots, M$$

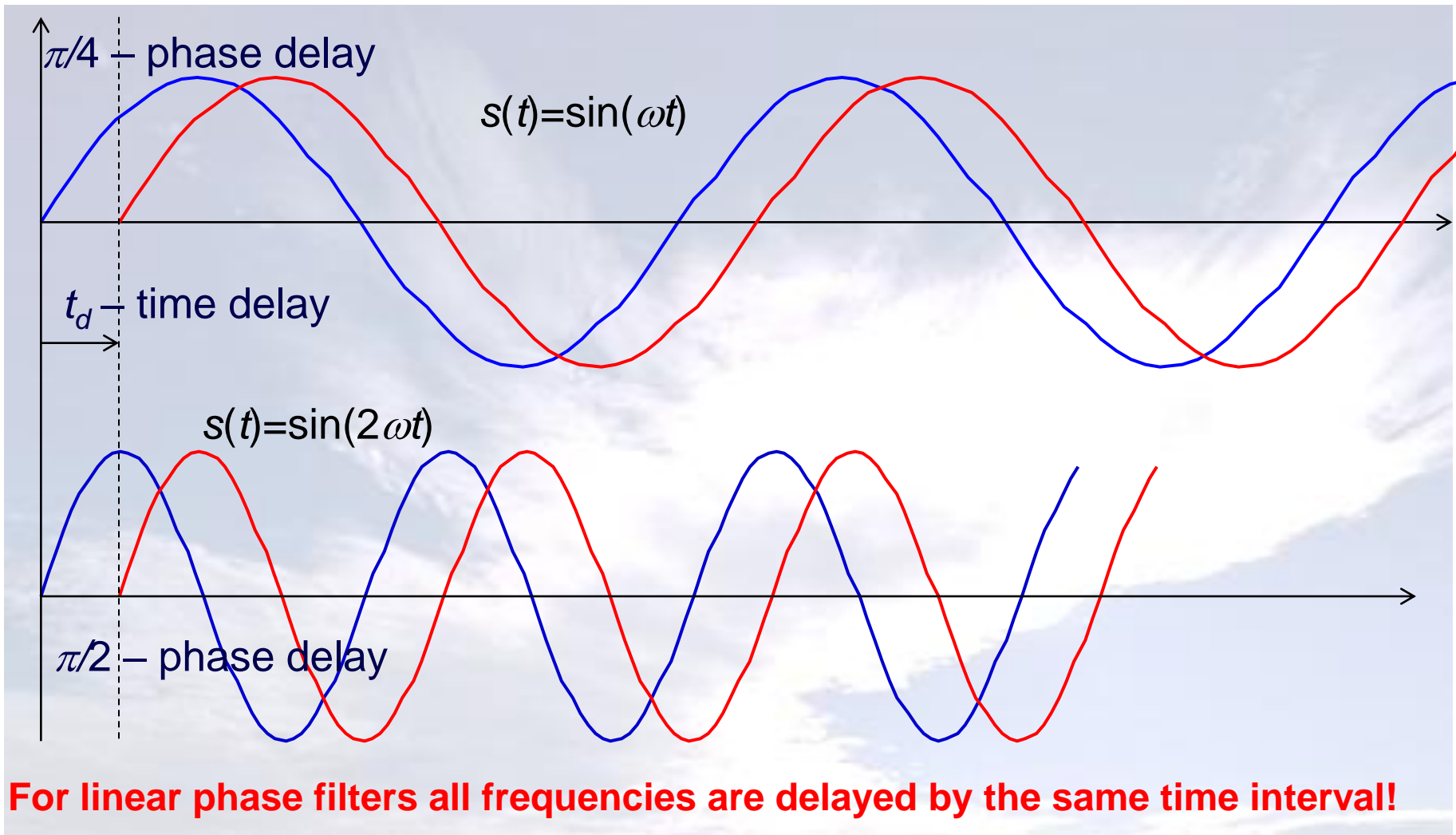
$$h(M - k) = -h(k) \quad \text{for odd or even } M$$

What is the advantage of the linear filter phase characteristic?

! The delay of the signal on the output is independent on its frequency.



## Linear phase → constant delay of all frequencies

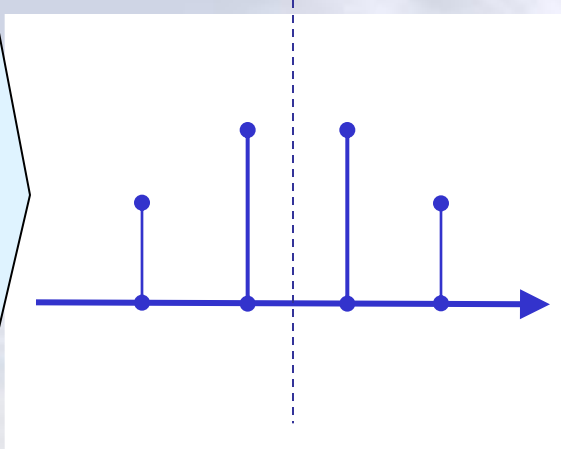


# Linear phase condition for FIR filters

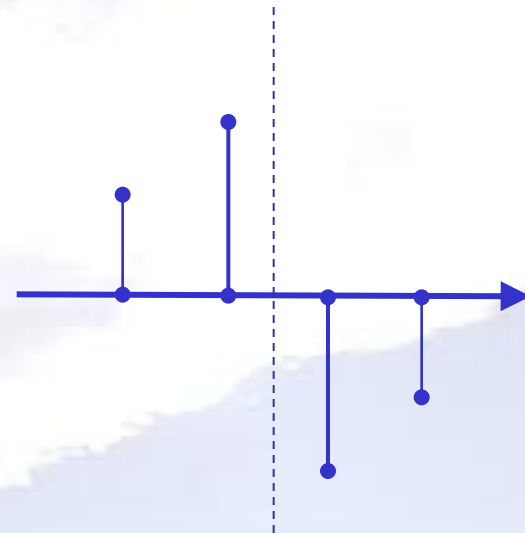
$$h(M - k) = h(k)$$

$$h(M - k) = -h(k)$$

for  $k = 0, 1, \dots, M$



*discrete time*

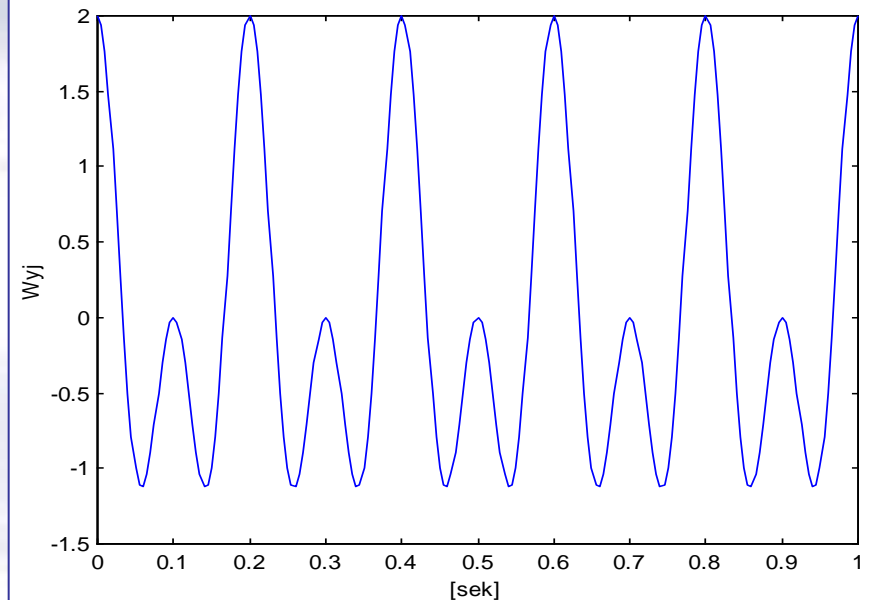
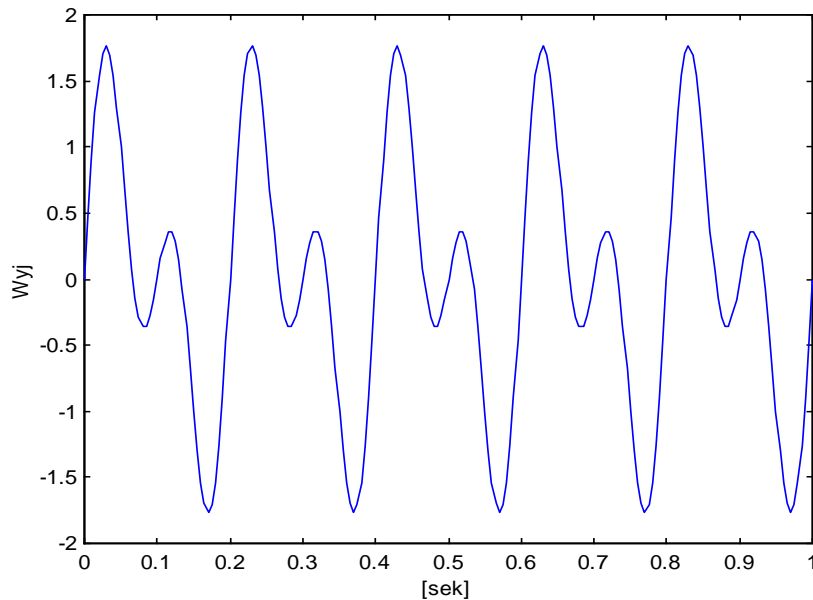


*discrete time*

# Phase distortion

$$x_{we} = \sin(10\pi t) + \sin(20\pi t)$$

$$x_{wy} = \sin\left(10\pi t + \frac{\pi}{2}\right) + \sin\left(20\pi t + \frac{\pi}{2}\right)$$

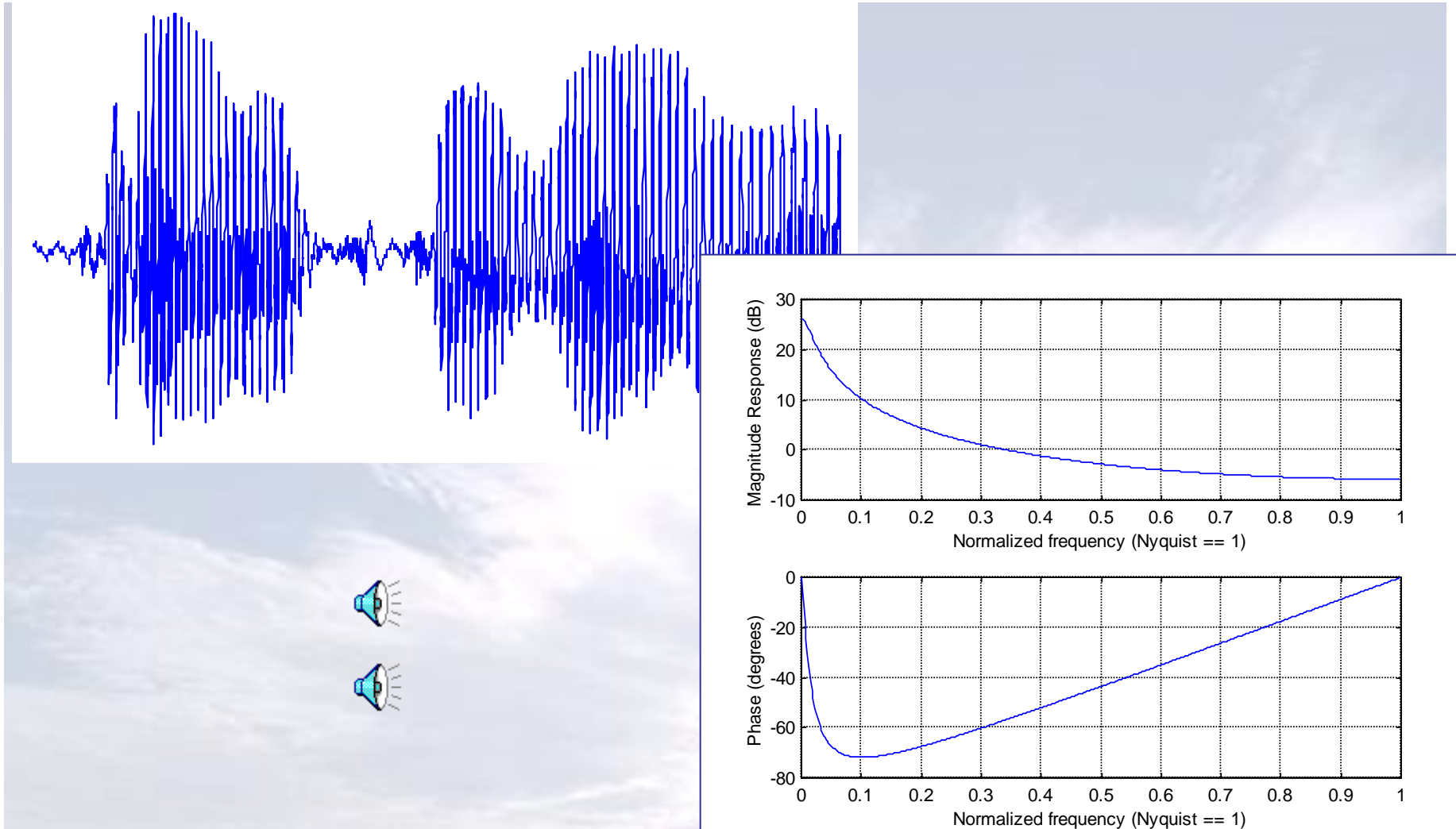


Nonlinear phase distortion

(eg. significant distortion of the acoustic signal)



# Phase distortion - example





## Other FIR filter examples

### Exercise cont.:

Determine the characteristics of the filters of the following impulse response:

- $h=0.25*[1 \ 2 \ 1]$  , so called von Hann filter (or Gaussian filter)
- $h=[1 \ -2 \ 1]$  ,  $\sim$  second derivative of the signal

# FIR filter design

## Examples

Low-pass from 0 to  $f$ :

```
>>> firwin(numtaps, f)
```

Use a specific window function:

```
>>> firwin(numtaps, f, window='nuttall')
```

High-pass ('stop' from 0 to  $f$ ):

```
>>> firwin(numtaps, f, pass_zero=False)
```

Band-pass:

```
>>> firwin(numtaps, [f1, f2], pass_zero=False)
```

Band-stop:

```
>>> firwin(numtaps, [f1, f2])
```

```
from scipy.signal import firwin
```

Where  $f$  is a normalised frequency  $f \in [0, 1]$ , where 1 corresponds to  $f_s/2$

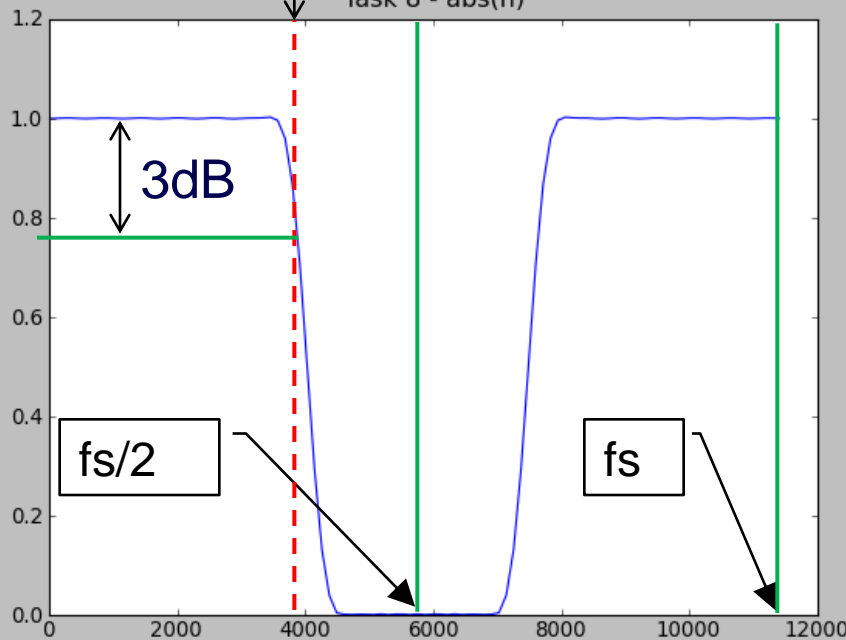
# FIR filter design - example

$M=40$

– no. of taps (filter order)

$$f_{\text{off}} = 0.7 \cdot f_s / 2$$

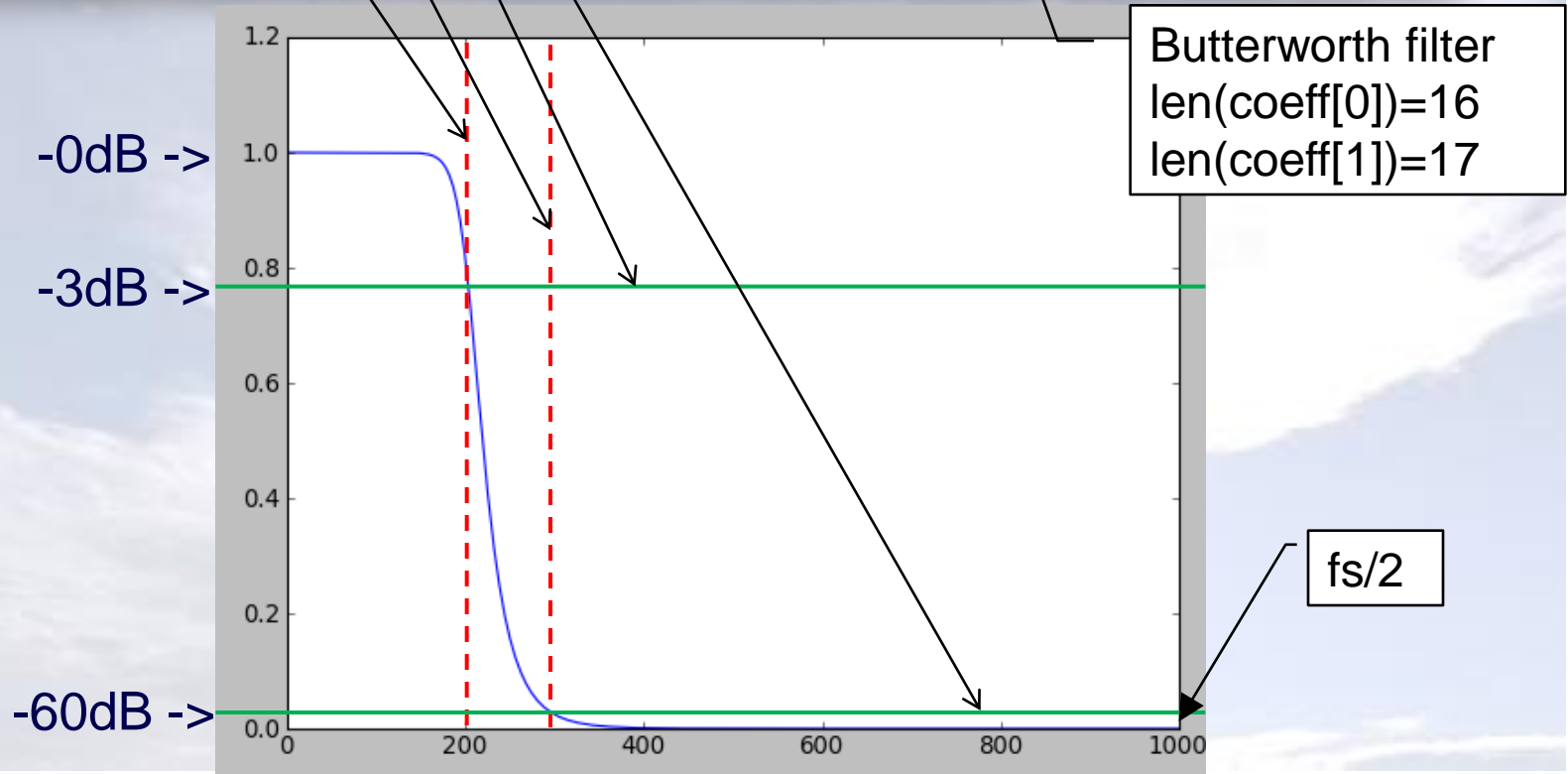
Task 8 - abs(h)



```
1 from pylab import figure, plot, show
2 from numpy import arange
3 from scipy.signal import firwin, freqz
4 b=firwin(40,0.7)
5 N=100
6 fs=1140
7 w,h=freqz(b,1,N, whole='True')
8 f=arange(0,fs,1.0*fs/N)
9 figure()
10 plot(f,abs(h))
11 show()
```

# IIR filter design - example

```
from scipy.signal import firwin, iirdesign
coeff=iirdesign(0.2,0.3,3,60,analog=0,ftype='butter',output='ba')
w,h=freqz(coeff[0],coeff[1])
plot(abs(h))
```



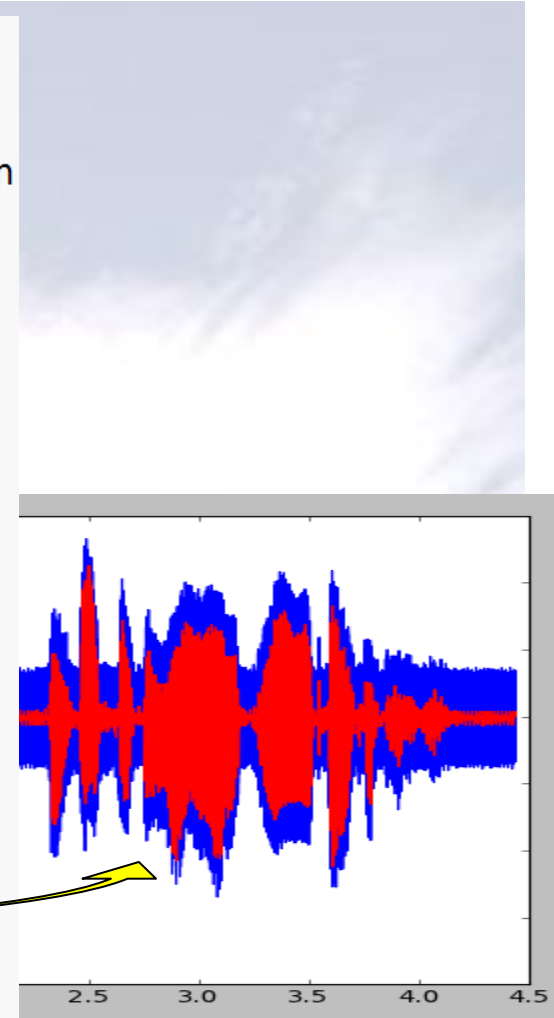


# Filter implementation

```
1 from scipy.io.wavfile import read as read_wav
2 from numpy import array, arange
3 from pylab import show, plot, title, figure
4 from scipy.signal import lfilter, freqz, convolve, firwin
5
6 fs, audio = read_wav('noise_voice.wav')
7
8 fs=11400
9 N=50580
10 T=float(N)/fs #time period definition
11 t=arange(0,T,1.0/fs)
12
13 figure(1)
14 plot(t,audio) #plot the noisy audio signal
15
16
17 b=firwin(40,0.7) #coefficients of the low-pass FIR filter
18
19 y = lfilter(b,[1],audio) # filtering of the audio signal
20
21
22 plot(t,y,'r') #plot the filtered audio signal
23 show()
```

**a[0]=1**

Diagram illustrating the filter implementation process. A yellow arrow points from the `lfilter` function call in line 19 to the plot of the filtered audio signal in line 22. A black arrow points from the `a[0]=1` box to the `[1]` argument in the `lfilter` function call.



# IIR and FIR filters comparison

## FIR

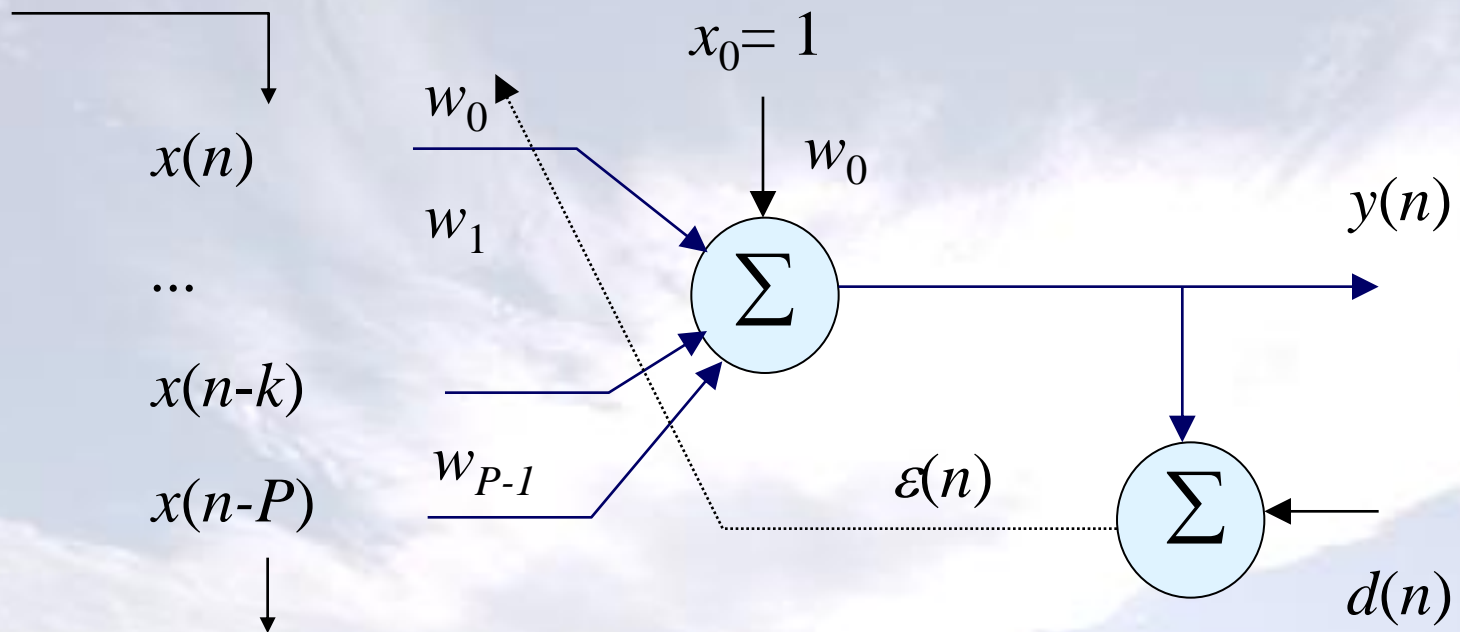
- Stable (from definition)
- Simple design
- condition for linear phase easy to satisfy
- Steep characteristic possible for very high orders of the filter
- Finite precision of filter coefficients is not a significant problem

## IIR

- Can be unstable
- Complex design
- Nonlinear phase
- Very steep characteristic possible for low orders of the filter
- Problems with implementation due to finite precision of filter coefficients.

# Adaptative filters

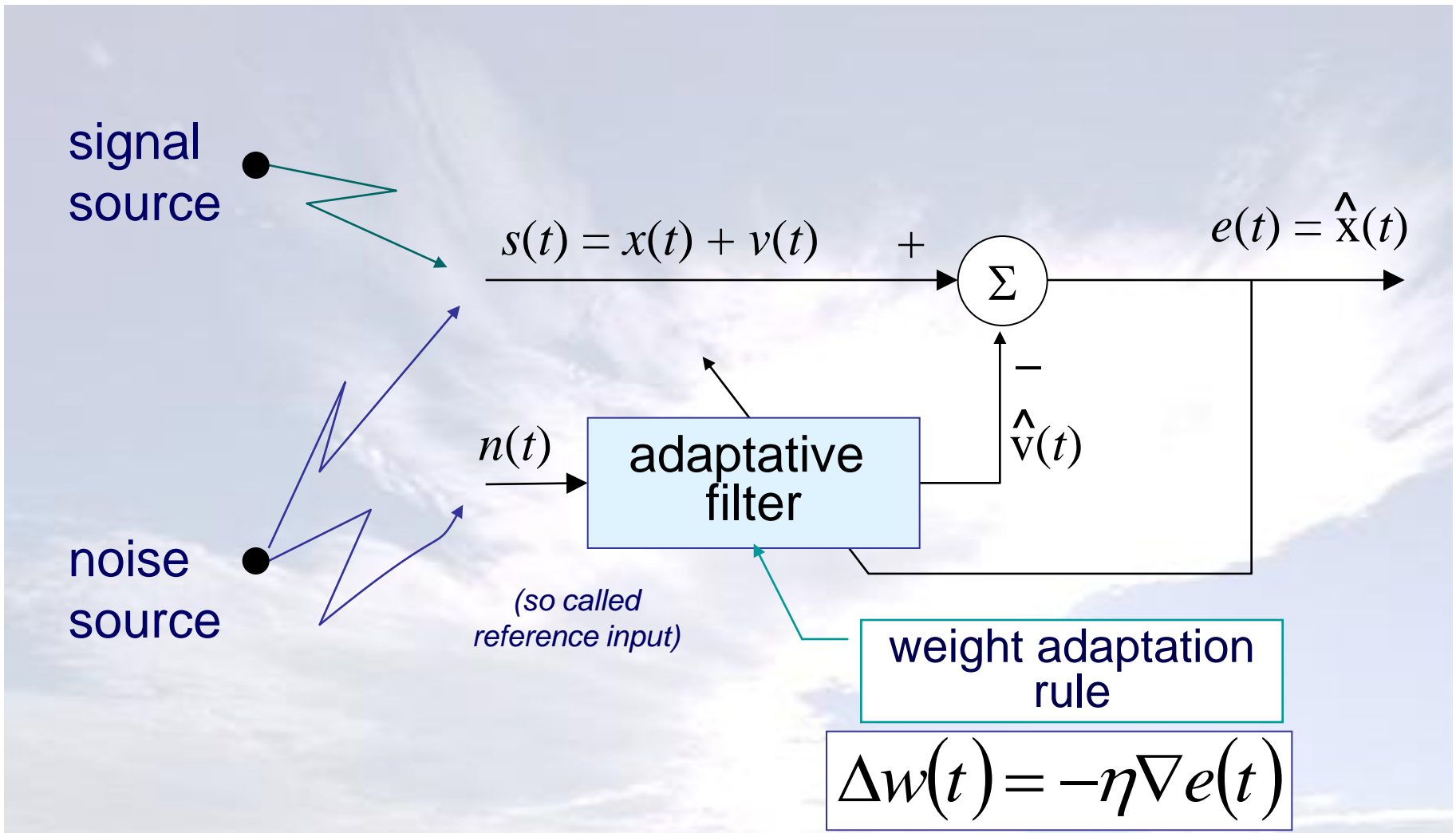
*Successive signal samples*



Target function:

$$E[d(n) - \mathbf{w}^T \mathbf{x}] = [(d(n) - y(n))^2] = E[\varepsilon^2] \rightarrow \min$$

# Adaptative noise reduction

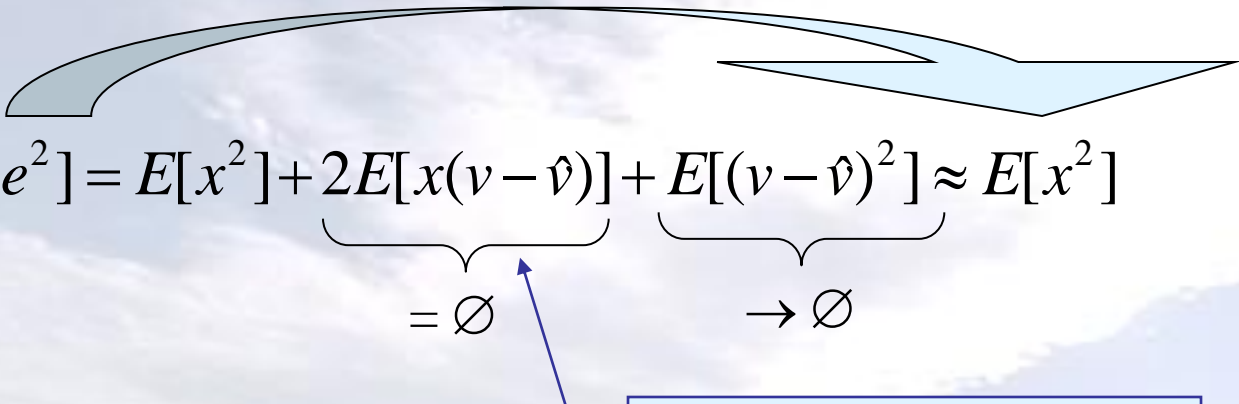




# Adaptative noise reduction

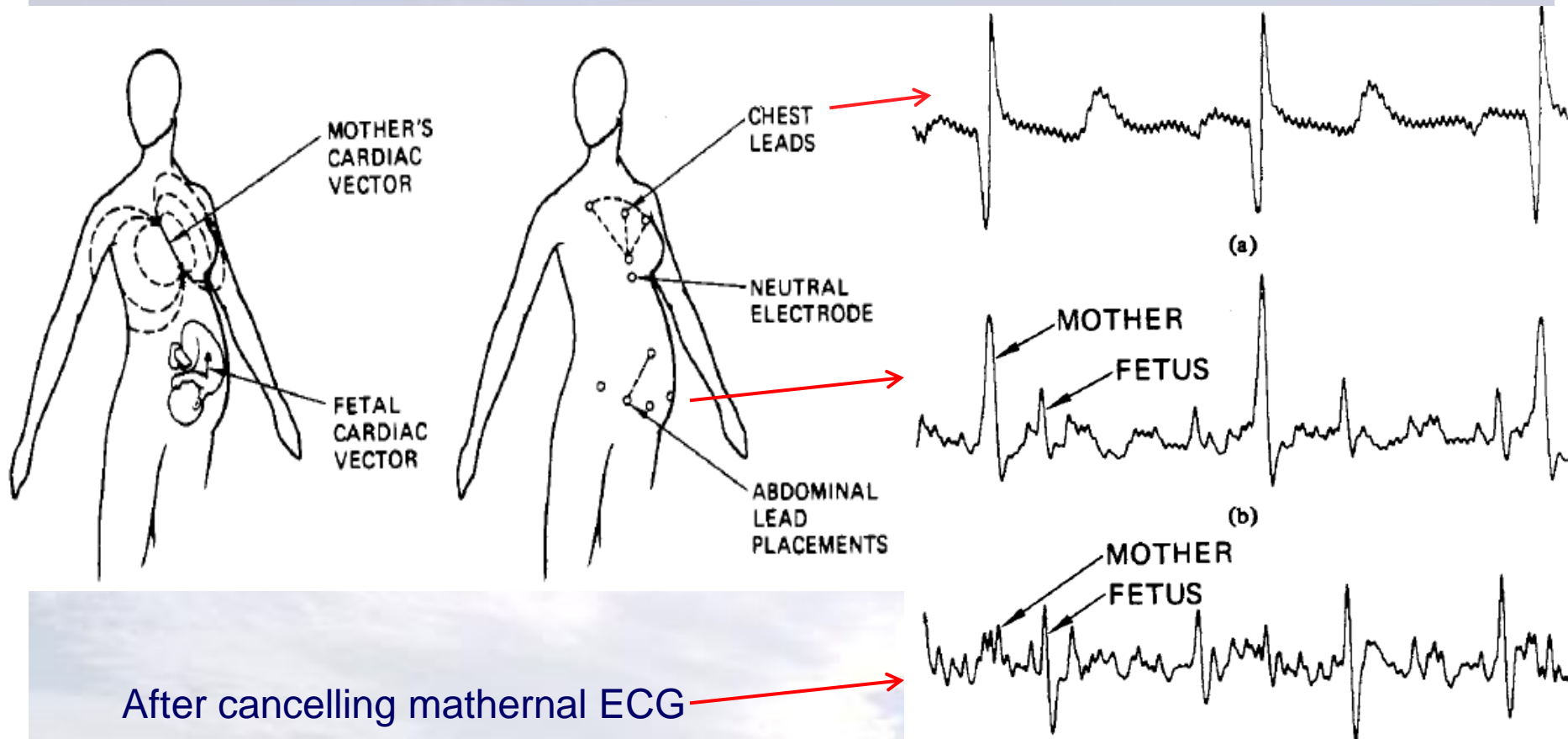
Minimal  $e(t)$  corresponds to the most effective (according to the minimum mean square error) noise reduction:

$$e(t) = s(t) - \hat{v}(t) = x(t) + v(t) - \hat{v}(t)$$


$$E[e^2] = E[x^2] + \underbrace{2E[x(v - \hat{v})]}_{= \emptyset} + \underbrace{E[(v - \hat{v})^2]}_{\rightarrow \emptyset} \approx E[x^2]$$

signal and noise are  
not correlated

# Adaptative noise reduction - application



After cancelling maternal ECG

Source: Widrow, B. ; Glover, J.R., Jr. ; McCool, J.M. ; Kaunitz, J. ; Williams, C.S. ; Hearn, R.H. ; Zeidler, J.R. ; Eugene Dong, Jr. ; Goodlin, R.C., **Adaptive noise cancelling: Principles and applications**, Proceedings of the IEEE Volume: 63 , Issue: 12, 1975

# Adaptative filtering application

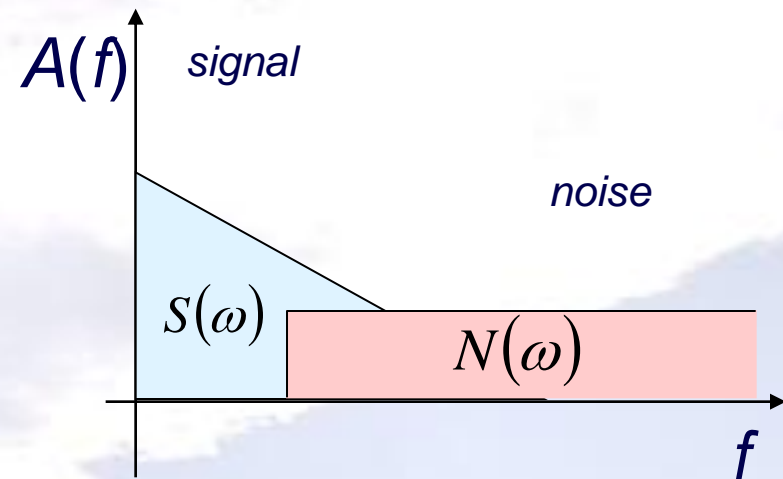
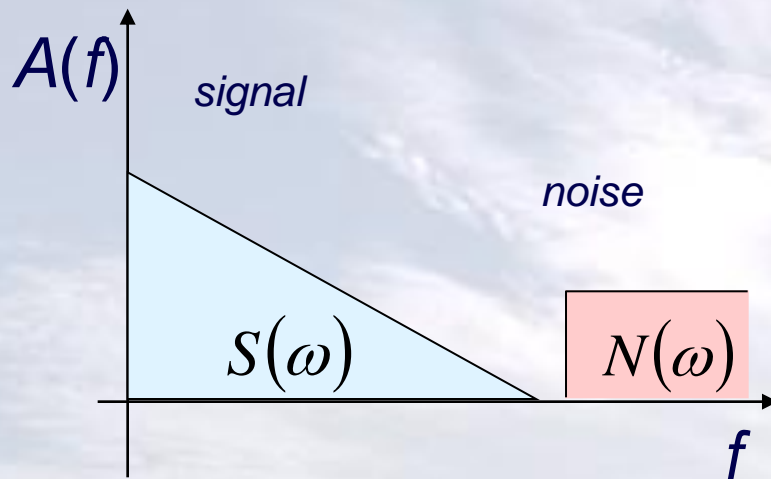
**Adaptative filters are mainly applied to filtering of nonstationary signals, eg:**

- In reduction of noise from the mains network and from electrical surgical instruments ( $f \sim 120$  Hz)
- In reduction of the energy of the ECG signal of a mother during registration of ECG signal of a fetus
- As a prediction model of biological signals in detection of their disturbances (eg. Ventricular fibrillation detection → implantation of defibrillators)

# Synchronous averaging of the signal

Lowpass filters are not effective for smoothing the signals in which the noise frequency band overlaps the significant frequency components of the signal.

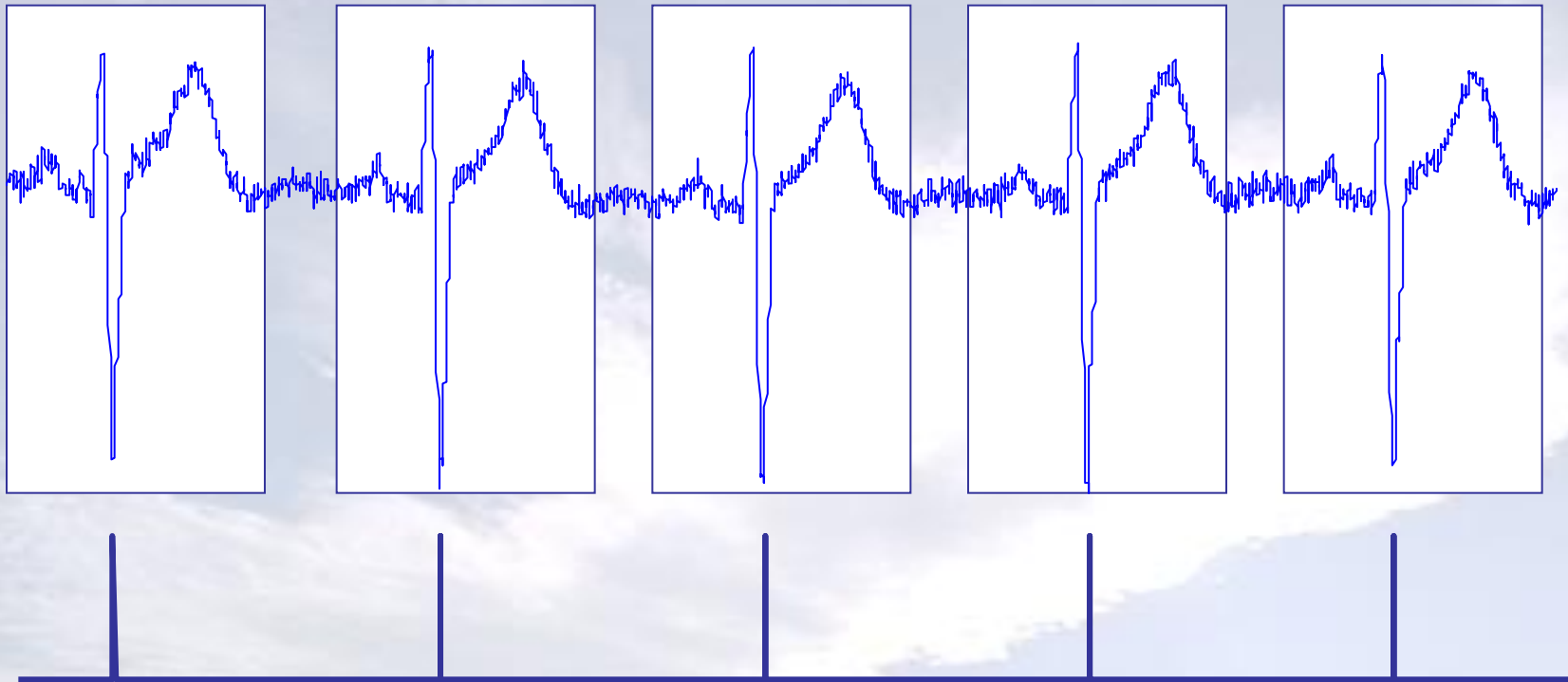
$$x(t) = s(t) + n(t)$$



Frequency spectra of the signal and noise

# Synchronous averaging of the signal

## Idea of synchronous averaging



*Synchronizing signal*



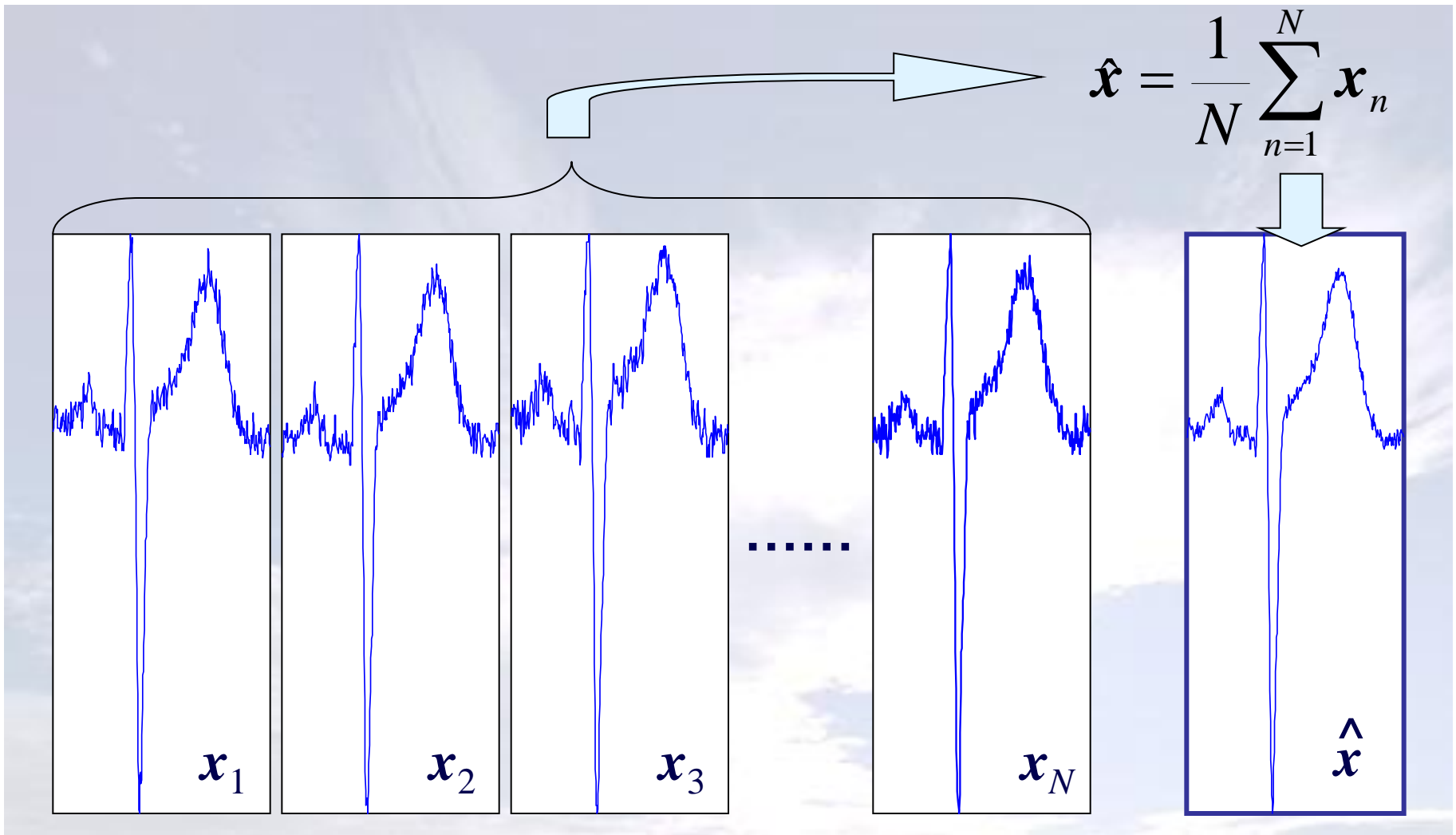
# Synchronous averaging of the signal

Synchronous averaging of the signal is efficient under the following conditions:

- deterministic components of the signal should occur periodically (not necessarily at regularly spaced intervals)
- disturbance signal should be a random signal, uncorrelated with the deterministic components of the signal.
- there should be a possibility to detect signal features necessary for synchronization of the successive cycles.



# Synchronous averaging of the signal



# Synchronous averaging of the signal

Standard deviation of the signal:

$$\sigma_s$$

Standard deviation of noise:

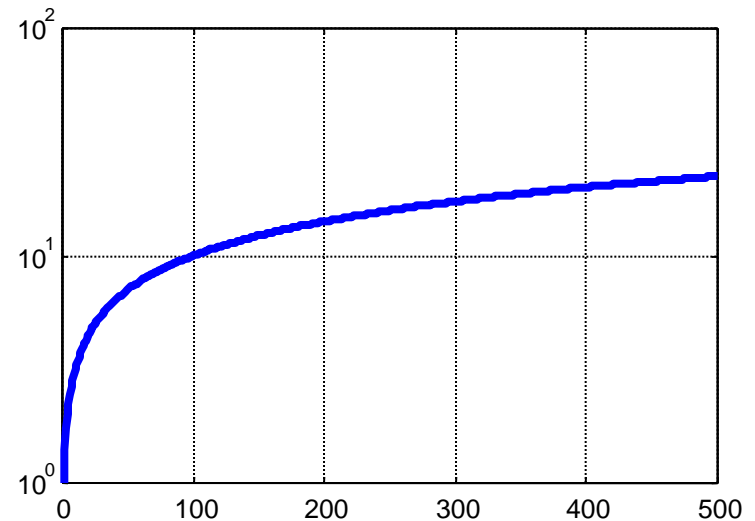
$$\sigma_n$$

Signal to noise ratio:

$$SNR = \frac{\sigma_s}{\sigma_n}$$

After  $N$  averagings:

$$SNR_N = \sqrt{N} \frac{\sigma_s}{\sigma_n}$$



SNR improvement  
after  $N$  averagings:

$$\sqrt{N}$$

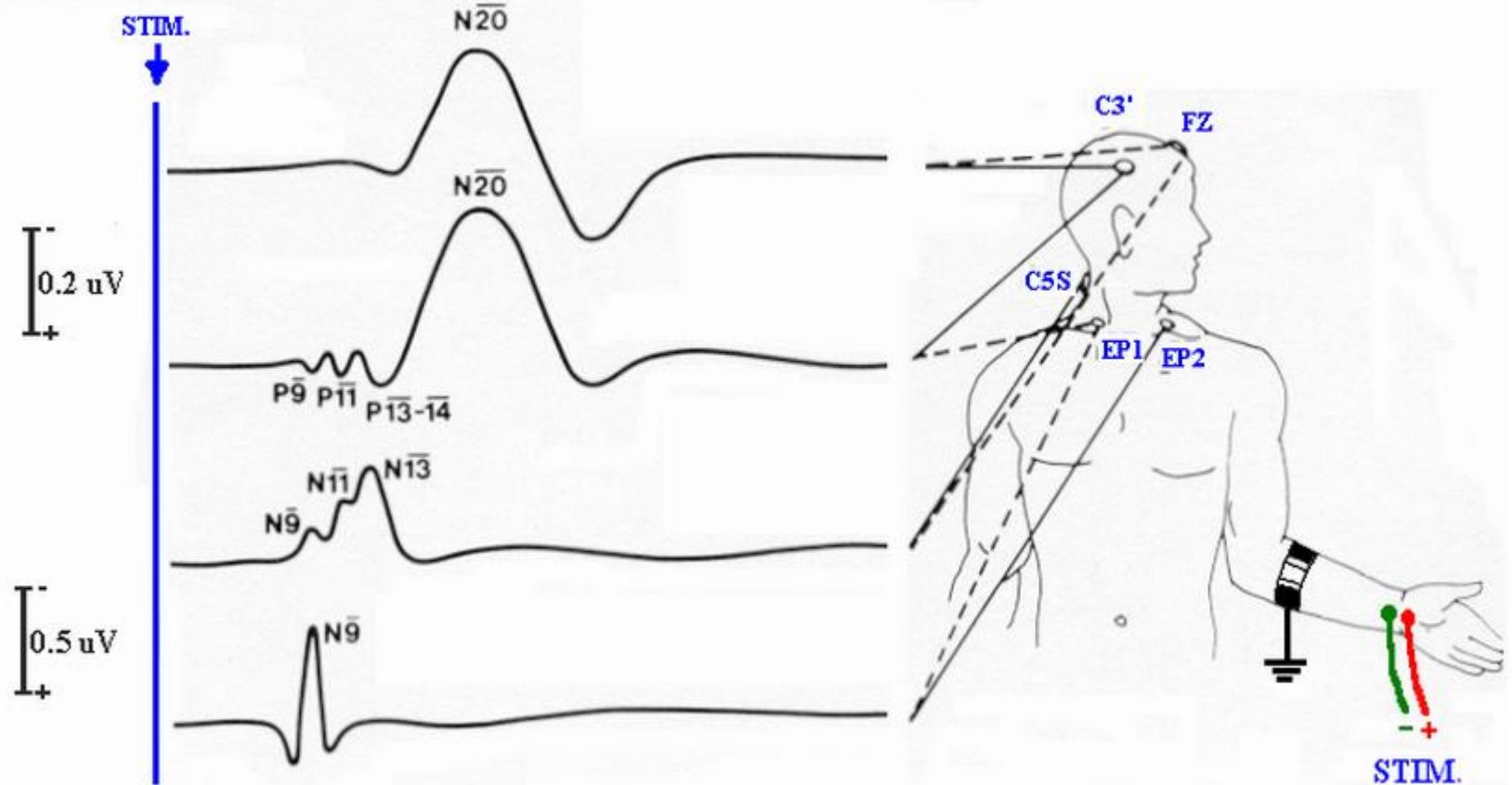


# Synchronous averaging of the signal

## Applications:

- sub-noise signal detection ( $\sigma_s \ll \sigma_n$ )  
(telecommunications)
- EEG electric evoked potentials analysis, ie. Analysis of potentials of the **several microvolts** amplitude, generated within the brain due to periodic stimulation by: light (**visual evoked** potentials), sound (**auditory evoked** potentials) or touch (**touch-evoked** potentials).

# Touch-evoked potentials



M. F. El-Bab, COGNITIVE EVENT RELATED POTENTIALS DURING A LEARNING TASK, PhD, University of Southampton, UK.



# Median filtering

**Median** – the middle element of the orderly sequence, eg.:

$$x(n) = \{1, 5, -7, 101, -25, 3, 0, 11, 7\}$$

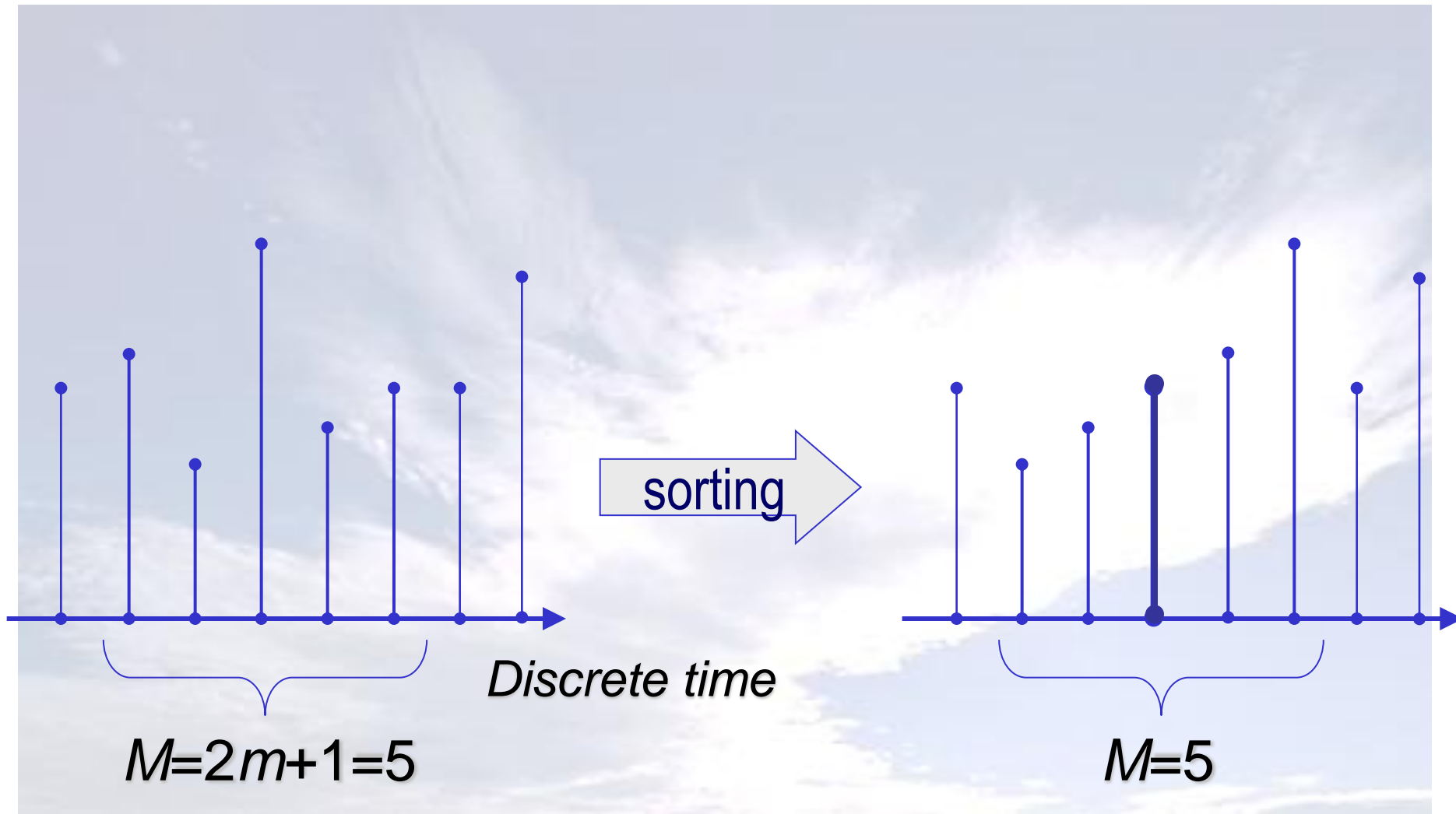
Sequence sorting:

$$x_s(n) = \{-25, -7, 0, 1, \mathbf{3}, 5, 7, 11, 101\}$$

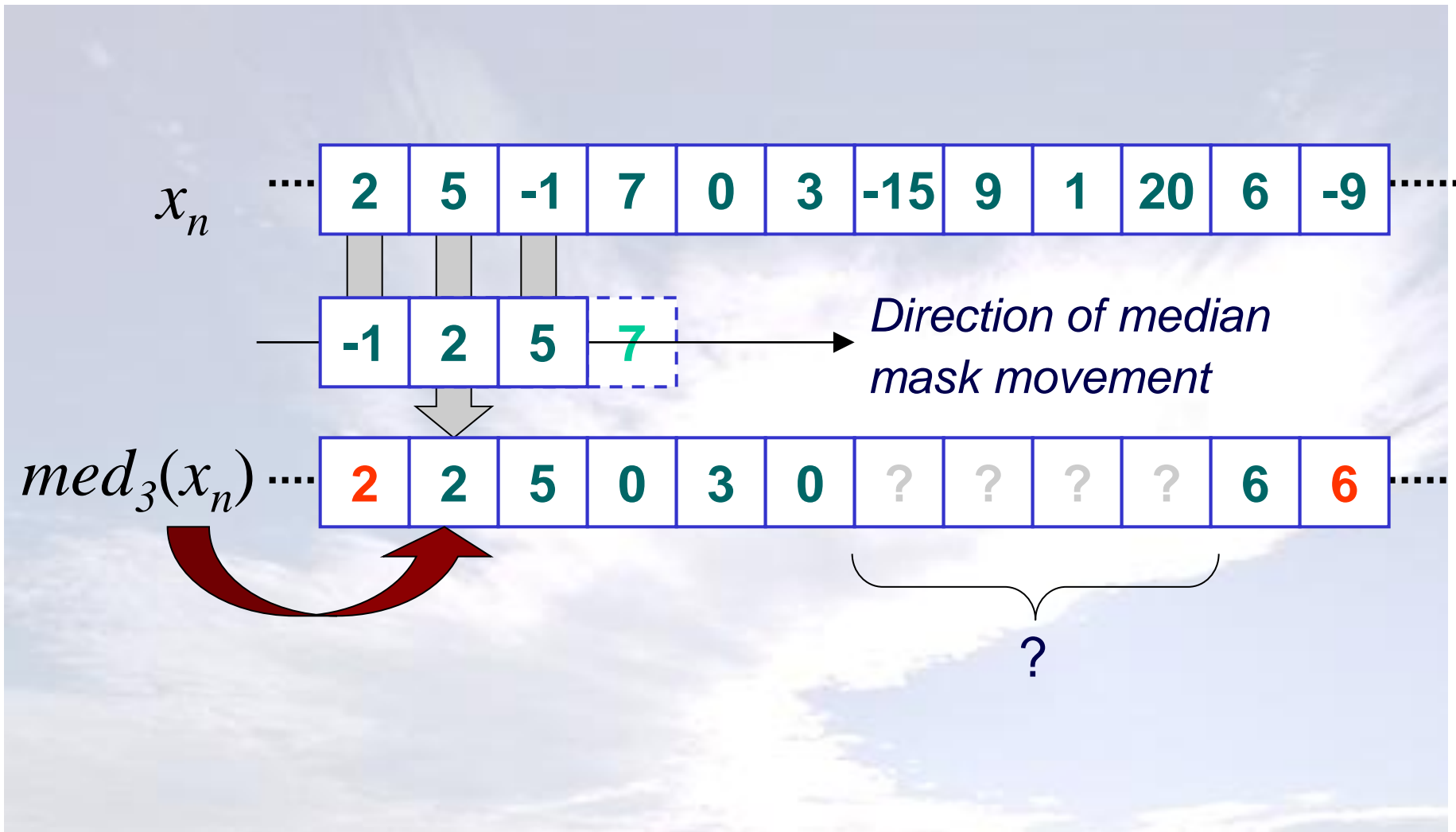
*Middle element*



# Median filtering of signals

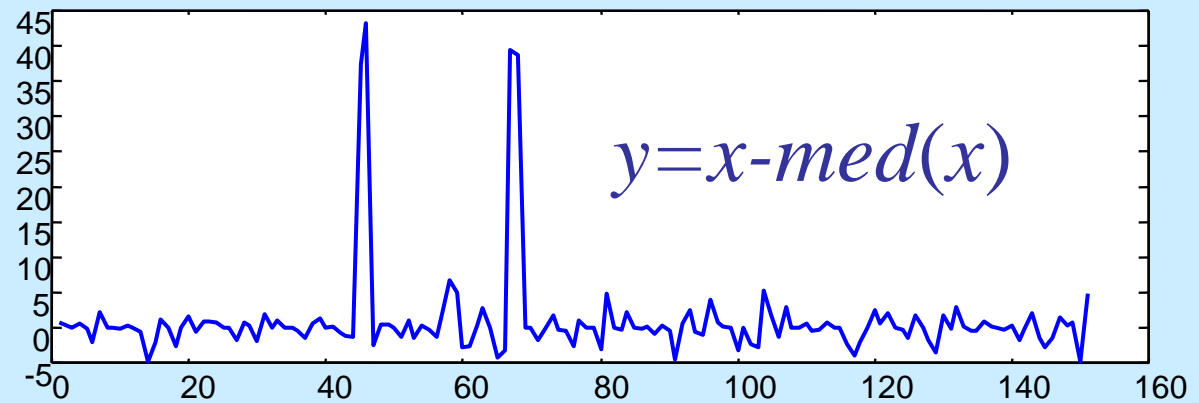
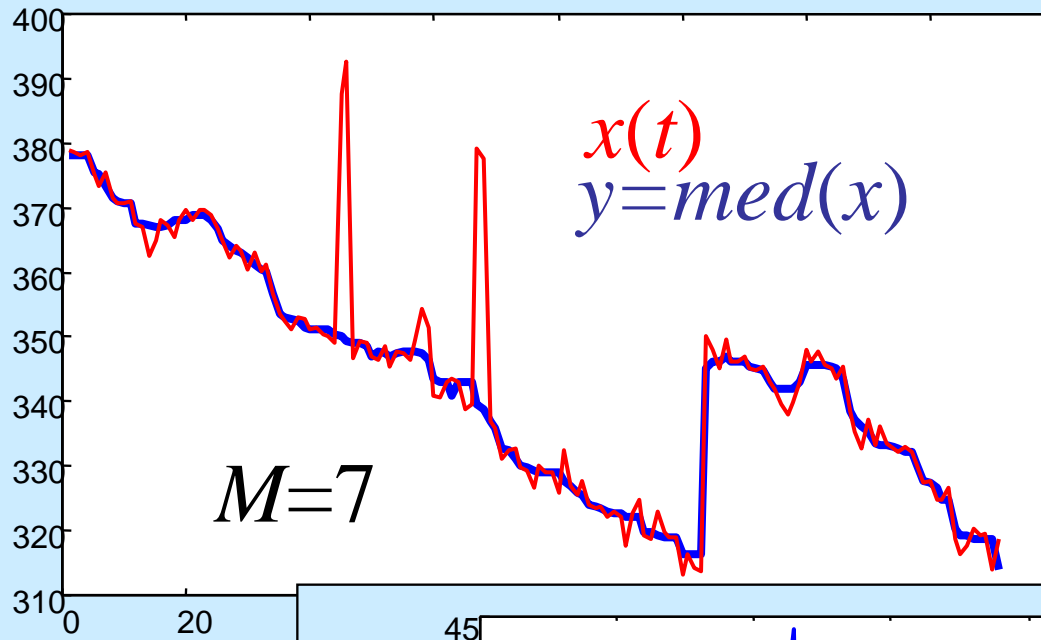


# Median filtering of signals





# Median filtering - example





# Digital filters - summary

1. Idea of filtering
2. Frequency characteristic of a filter
3. Ideal filters
4. Implementable filters
5. Difference equation
6. FIR and IIR filters
7. Linear phase and phase distortion
8. Filter design and signal filtering
9. Adaptive filtering
10. Median filtering (→nonlinear filtering)





**KAPITAŁ LUDZKI**  
NARODOWA STRATEGIA SPÓJNOŚCI

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nowoczesna oferta edukacyjna i wzmacniania zdolności  
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