frontiers in PSYCHOLOGY

GENERAL COMMENTARY
published: 09 July 2012
doi: 10.3389/fpsyg.2012.00233

Filter effects and filter artifacts in the analysis of electrophysiological data

Andreas Widmann* and Erich Schröger

Digital filter design for electrophysiological data – a practical approach

Andreas Widmann[1*], Erich Schröger[1], and Burkhard Maess[2]

[1]*Cognitive and Biological Psychology, University of Leipzig, Germany*
[2]*Max Planck Institute for Human Cognitive and Brain Sciences, Leipzig, Germany*

Signal filtering

- Filtering is an **almost ubiquitous** step in the preprocessing of biosignals.

- It lies in the nature of this process itself that filtering might seriously change the **appearance** of the signals and **thereby affect the results obtained**.

- Filters **improve** the signal-to-noise ratio but also **introduce** signal distortions

Signal filtering

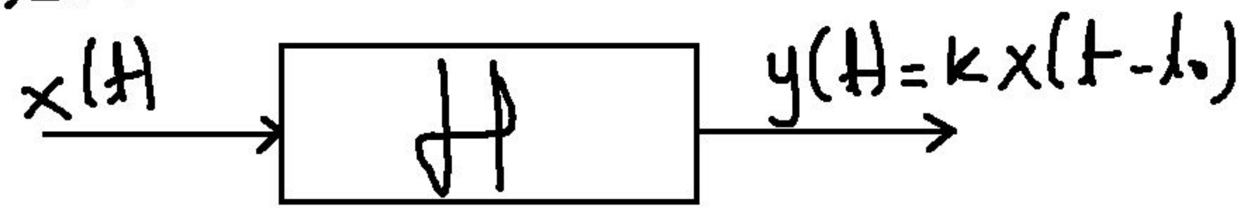
- **Temporal filtering or frequency filtering** (in contrast to spatial and other types of filtering) refers to the attenuation of signal components of a particular frequency (band).
- The common rationale behind filtering in general is to **attenuate noise** in the recordings, while preserving the signal (of interest).
- In some applications neither noise nor signal are clearly defined.

Signal filtering

- Typically, there is an **overlap** of signal components and noise components in the same frequency band.
- The temporal filters discussed here cannot separate signal from noise in the **same band**; they will simply attenuate everything in the targeted band.

Signal filtering

Sea:

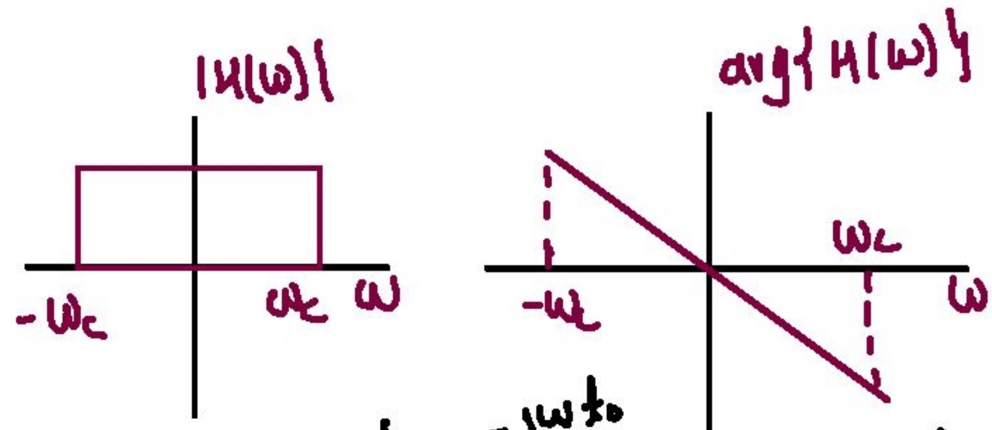


Con $x(t) \xleftrightarrow{\mathcal{F}} X(\omega)$

$$y(t) \xleftrightarrow{\mathcal{F}} Y(\omega) = k e^{-j\omega t_0} X(\omega)$$

$$\therefore \quad H(\omega) = \frac{Y(\omega)}{X(\omega)} = k e^{-j\omega t_0}$$

Signal filtering



$$H(\omega) = \begin{cases} k\,e^{-j\omega t_0} & -\omega_c \leq \omega \leq \omega_c \\ 0 & \text{e.o.c} \end{cases}$$

Cómo será $h(t)$? (Respuesta al impulso)

$$h(t) = \frac{k}{2\pi}\int_{-\omega_c}^{\omega_c} e^{j\omega t}\, e^{-j\omega t_0}\, d\omega$$

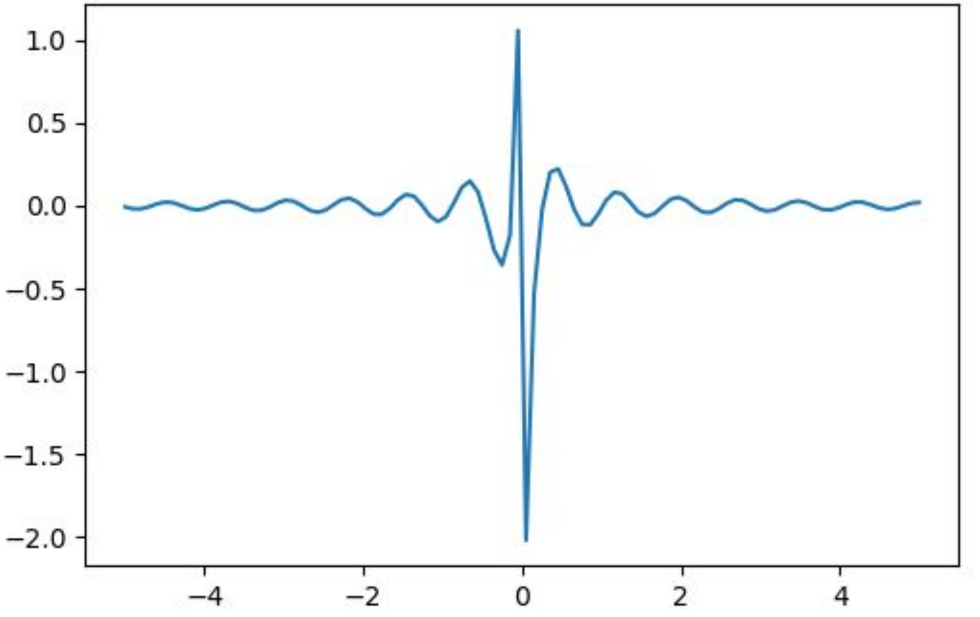
Signal filtering

- What happens with h(t)?

```
import numpy as np
import matplotlib.pyplot as plt

t = np.linspace(-5,5,100)
wc = 2*np.pi*50
t0 = 0.001
ft = (wc/np.pi)*np.sinc(wc*(t - t0))

plt.plot(t,ft)
plt.show()
```

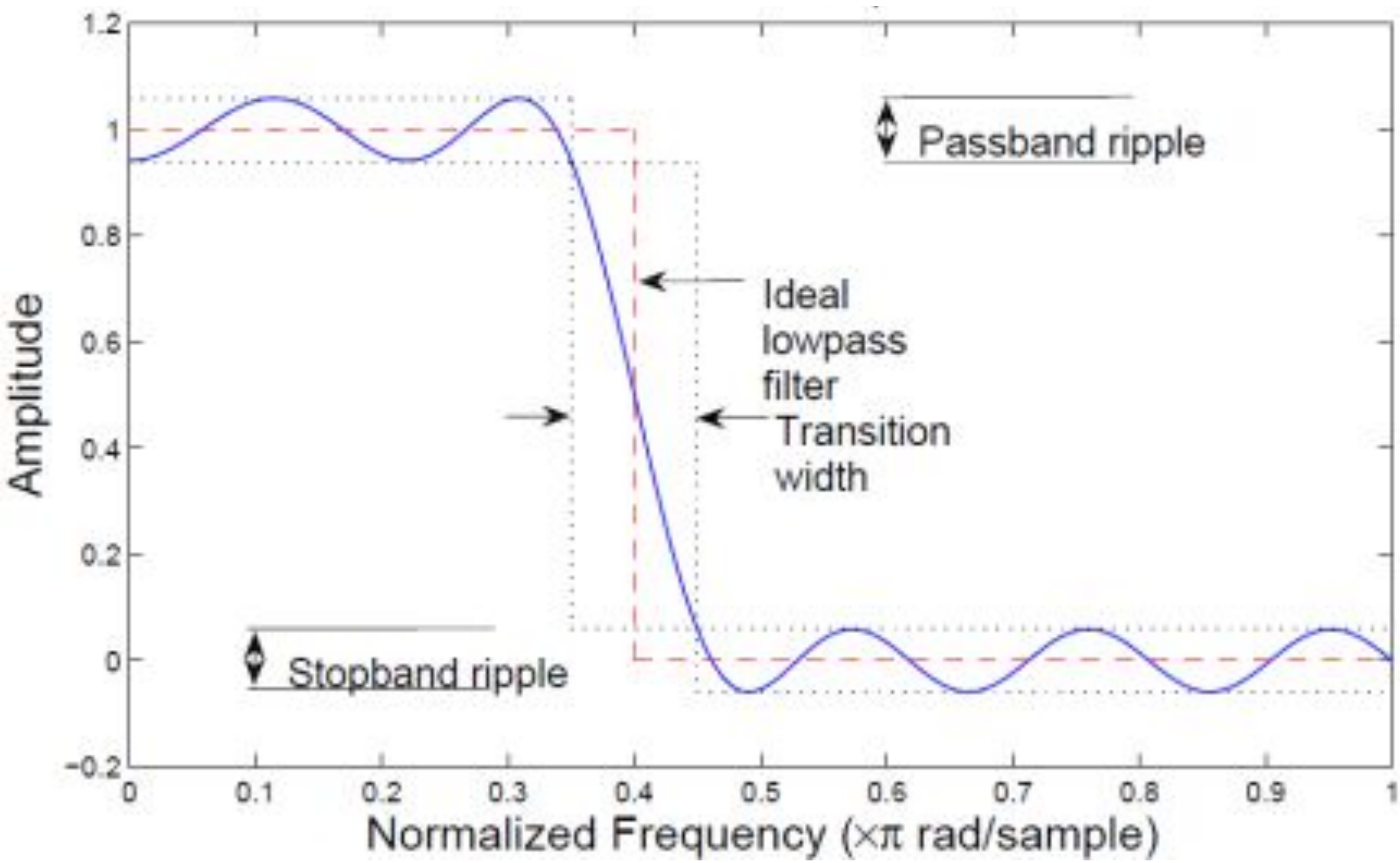


Signal filtering

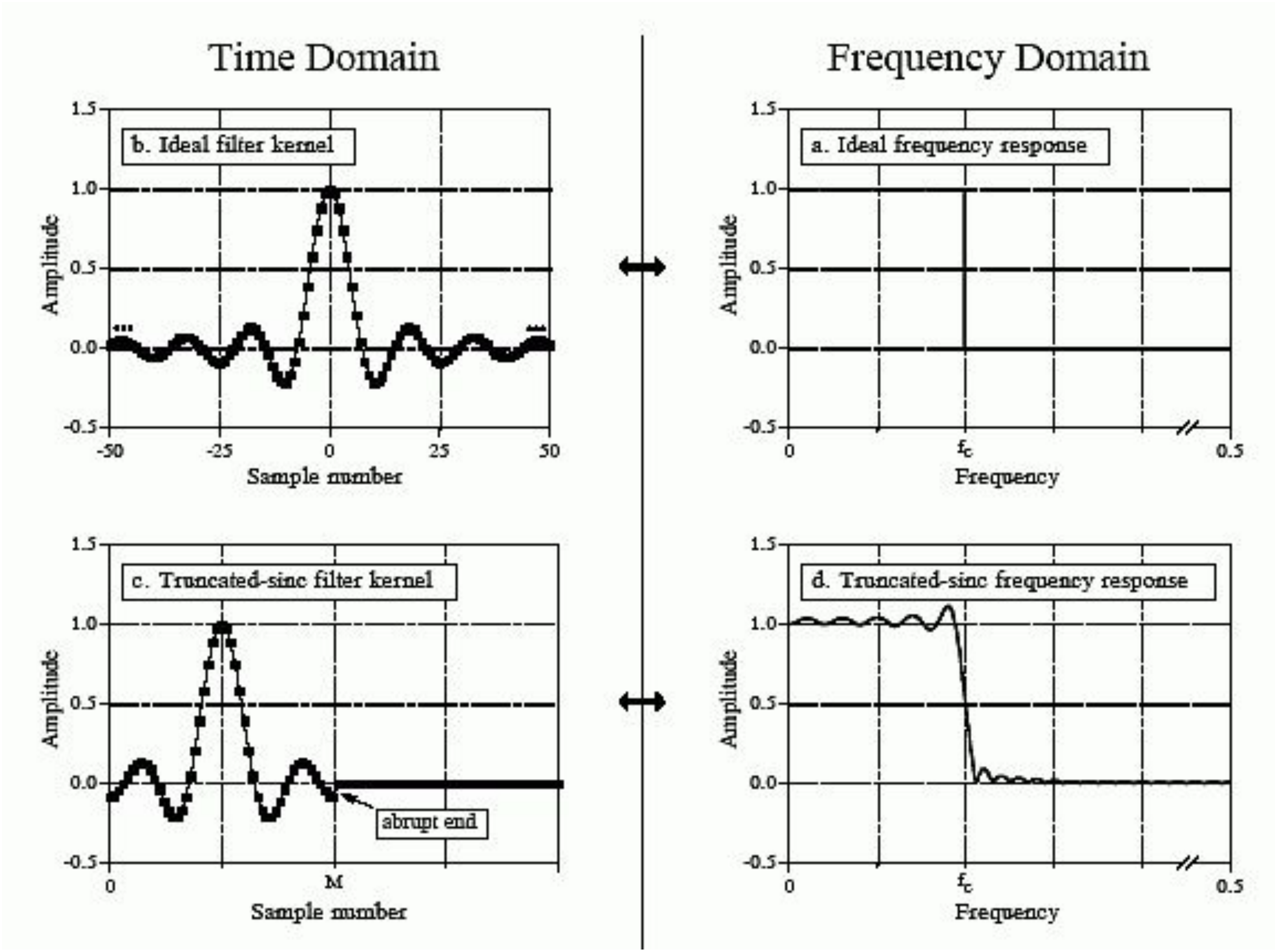
Causal filters

- $|H(w)|$ can be zero in some frequencies but can't be zero over any frequency band
- The amplitude of the band pass frequency can't be constant
- Magnitude and phase can't be independently manipulated

Signal filtering



Signal filtering



Signal filtering

Digital filters

- The digital filter bandwidth is limited by the sampling frequency. For the analog filter is limited by the amplifiers
- There is not saturation (although there are limitations by the number of bits used)
- Can be easily copied and transferred among systems
- Can be implemented using software or hardware
- There are not problems with tolerances and impedances
- The computers evolve faster than conventional electronics, so, better designs in digital filters can be reached

Signal filtering

Digital filters

FIR

$$H[z] = \frac{Y[z]}{X[z]} = B_0 + B_1 z^{-1} + B_2 z^{-2} + \ldots + B_M z^{-M}$$

$\{B_k\}$: coeficientes del filtro

M: orden del filtro

IIR

$$H[z] = \frac{B_0 + B_1 z^{-1} + B_2 z^{-2} + \ldots + B_M z^{-M}}{1 + A_1 z^{-1} + A_2 z^{-2} + \ldots + A_N z^{-N}}$$

Signal filtering

Digital filters

- FIR (Finite Impulse Response): Always stable

- IIR (Infinite Impulse Response): Need a lower order to reach the same specification of the FIR filter.

  Have lower ripple than the FIR filter
  Have non-linear phase.

  Can be inestable

  Accumulated rounding errors result in deviating filter responses.

Signal filtering

- Butterworth filters (IIR) have no passband and stopband ripple and have the shallowest roll-off near the cutoff frequency (the most relevant roll-off region) compared to the other commonly used (Chebyshev and elliptic IIR filters).

Signal filtering

- Despite IIR filters often being considered as computationally more efficient, they are recommended only when high throughput and sharp cutoffs are required.

- **For offline data analysis, however, throughput and computational time do not matter on modern computer hardware**.

Signal filtering

- The impulse response is also finite for IIR filters due to numerical precision, thus, **all relevant properties can also be implemented with FIR filters**.

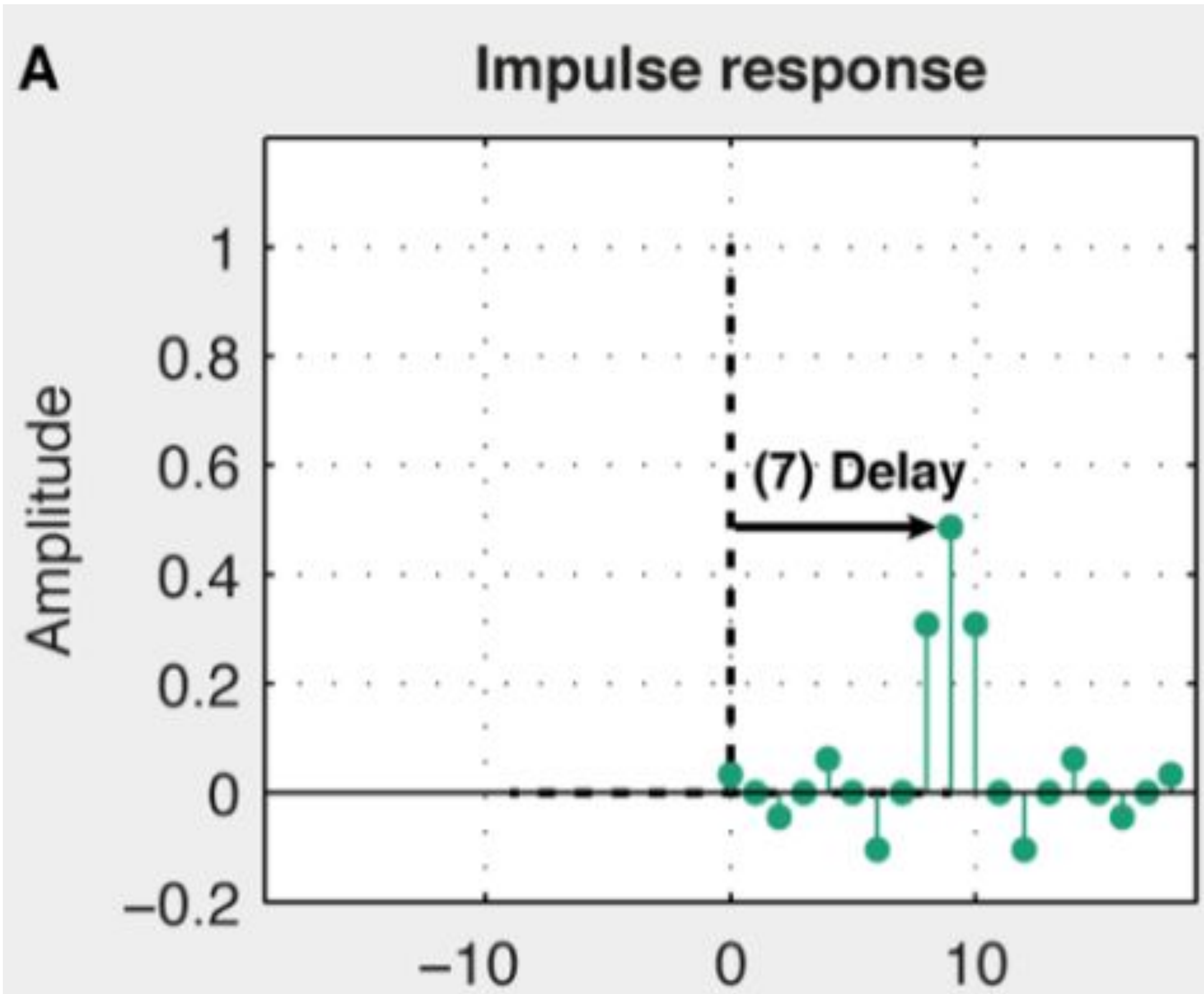
- FIR filters are easier to control, are always stable, have a well-defined passband, can be corrected to zero-phase without additional computations, and can be converted to minimum-phase.

Filter design – Filter responses

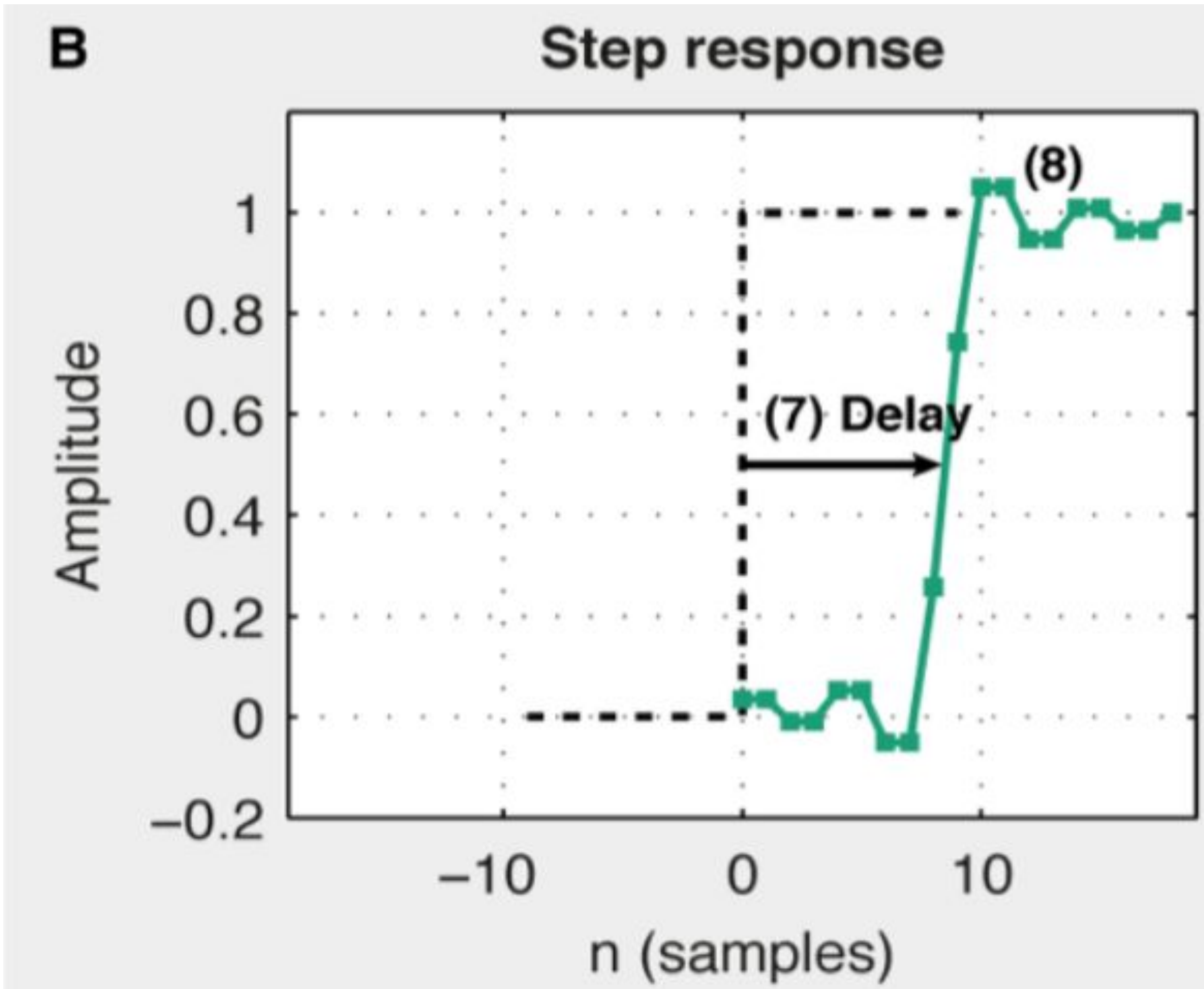
Filter design – Filter responses

- **As impulse and step signals have energy across the whole spectrum** they are excellent tools to evaluate possible filter distortions when filtering broadband complex signals.

Filter design – Filter responses

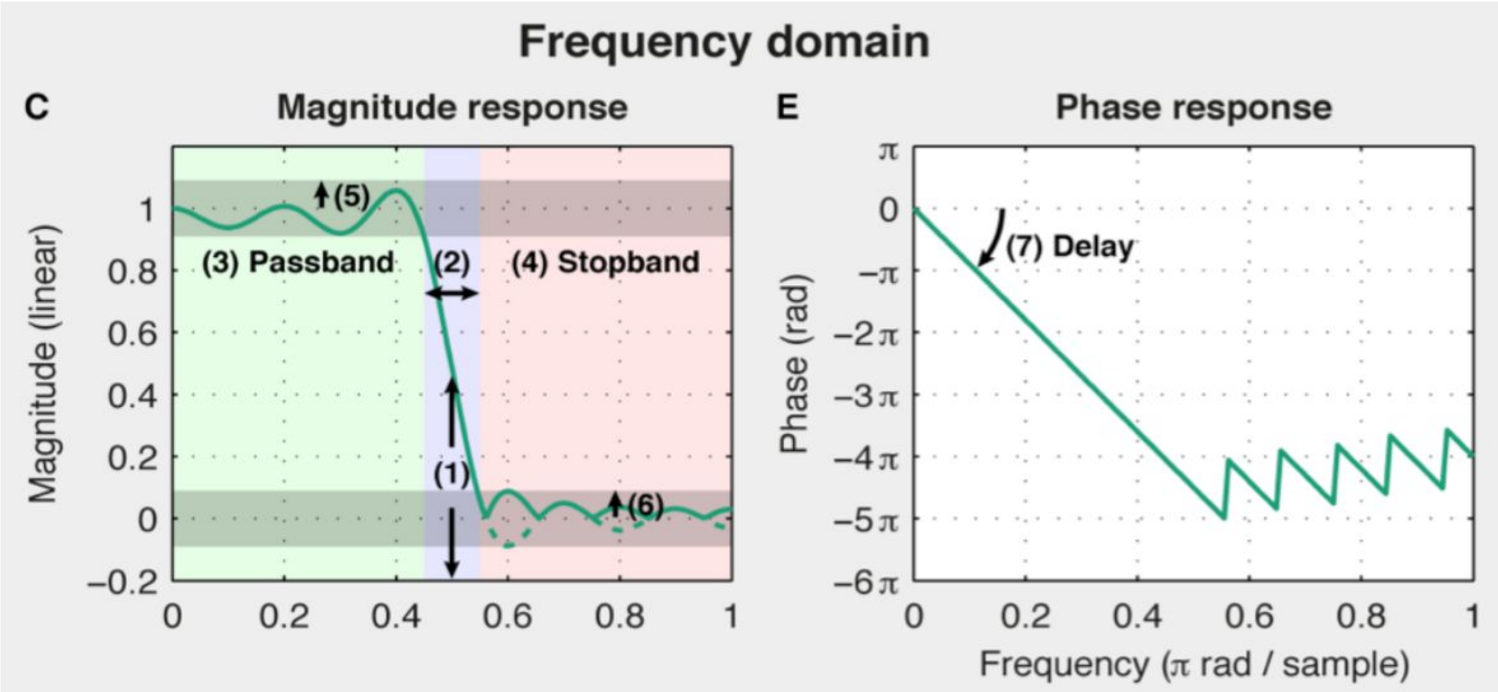


Filter design – Filter responses



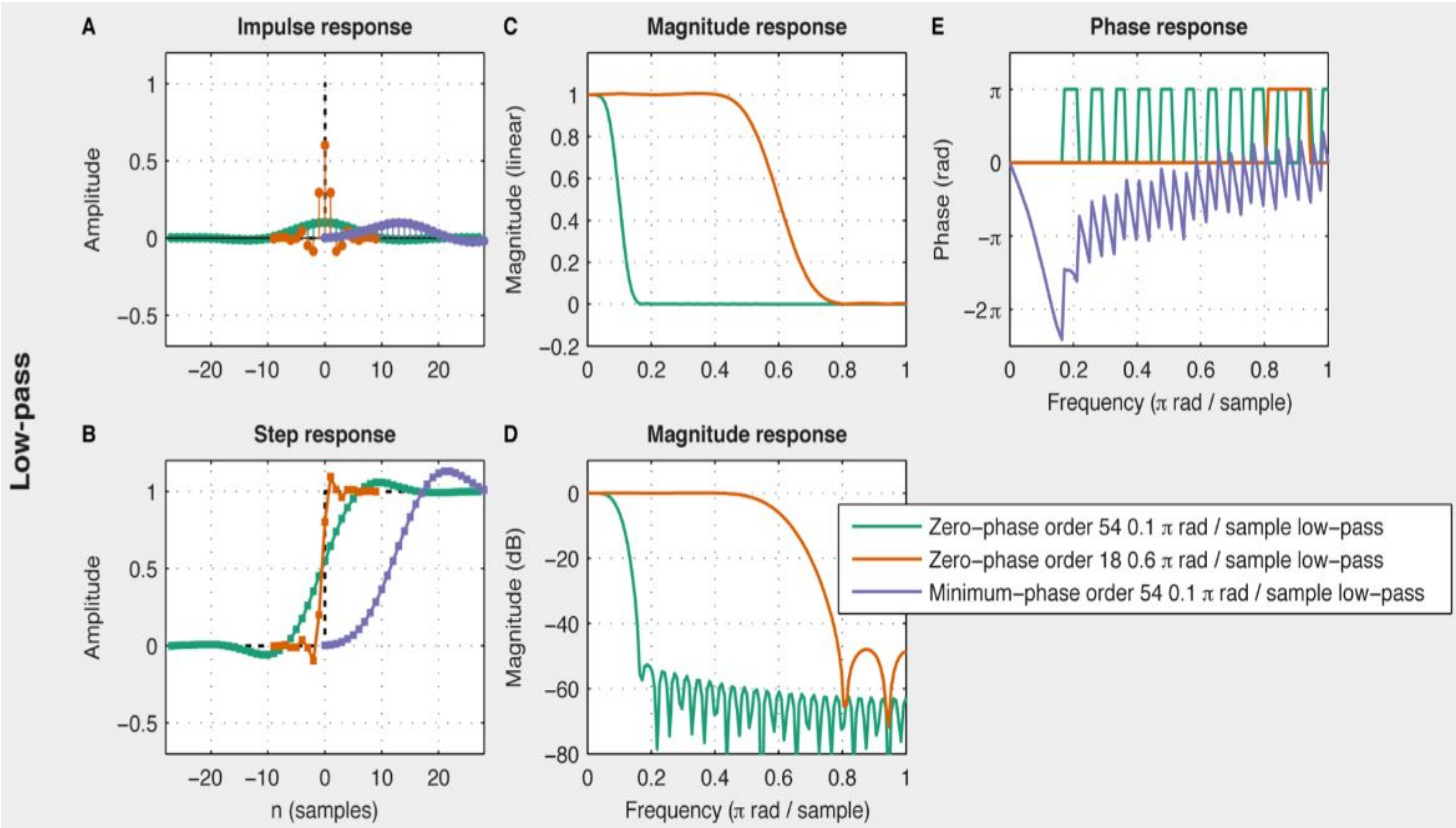
Filter design – Filter responses

- The frequency response is the Fourier transform of the impulse response and consists of two parts: **magnitude and the phase response**.

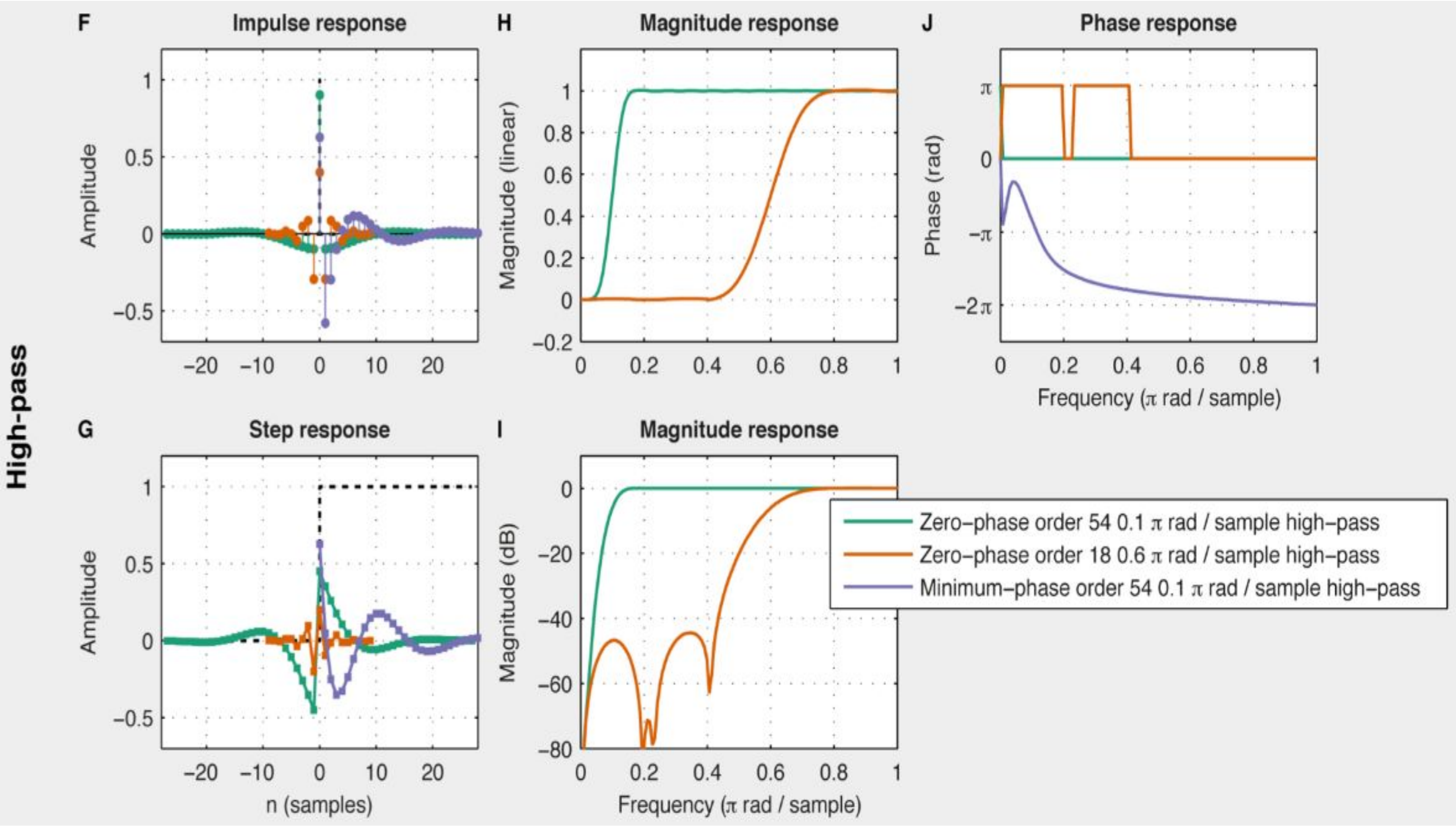


Filter design – Filter type

Filter design – Filter type

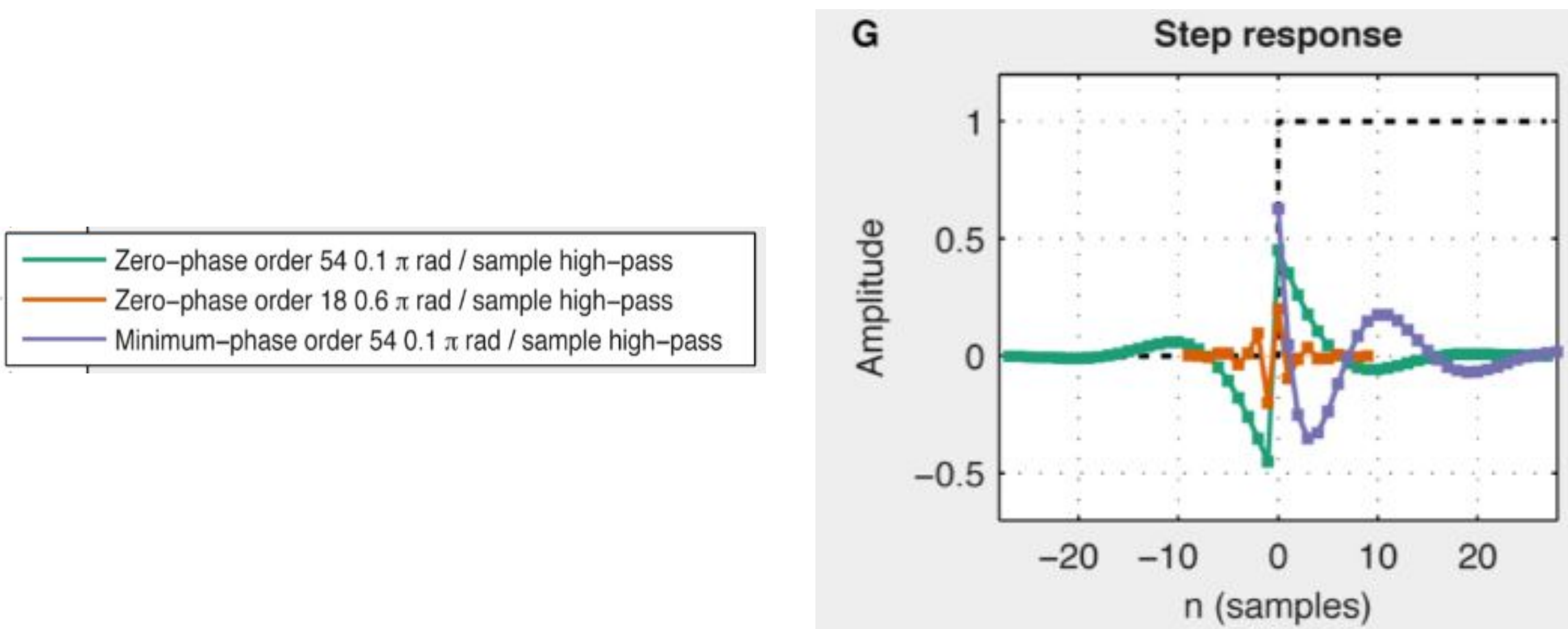


Filter design – Filter type



Filter design – Filter type

- Note that zero-phase filters introduce a **symmetric change in the signal around a step**. These types of filter **distortions** can easily be observed in the step response.



Filter design – Filter type

- A separate successive application of a steep **high-pass** and a shallow **low-pass** filter is often preferred over a band-pass filter.

- The use of digital **band-stop** filters is not recommended in biosignal research as they likely produce strong artifacts.

Filter design – Filter type

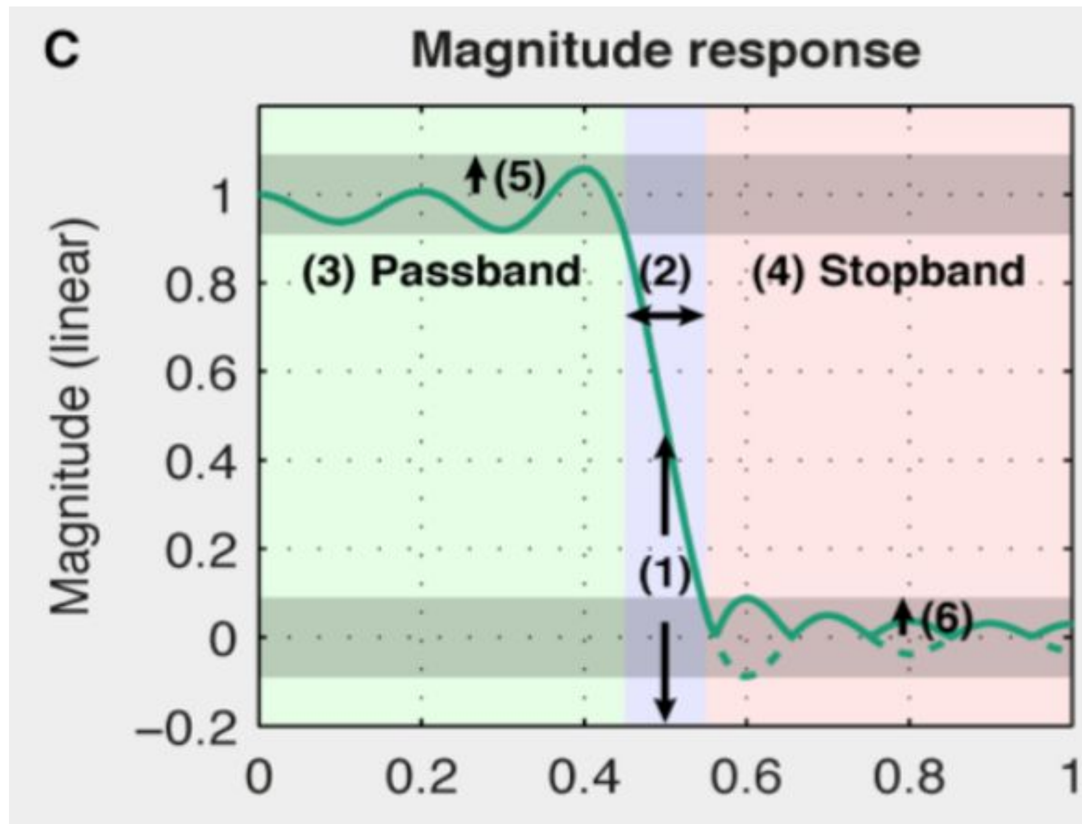
- Band-stop filters are almost exclusively used to suppress line (50/60 Hz) and should be replaced **by time domain regression-based approaches.**

- These approaches are superior due to the very high phase stability of line noise.

Filter design – Cutoff Frequency

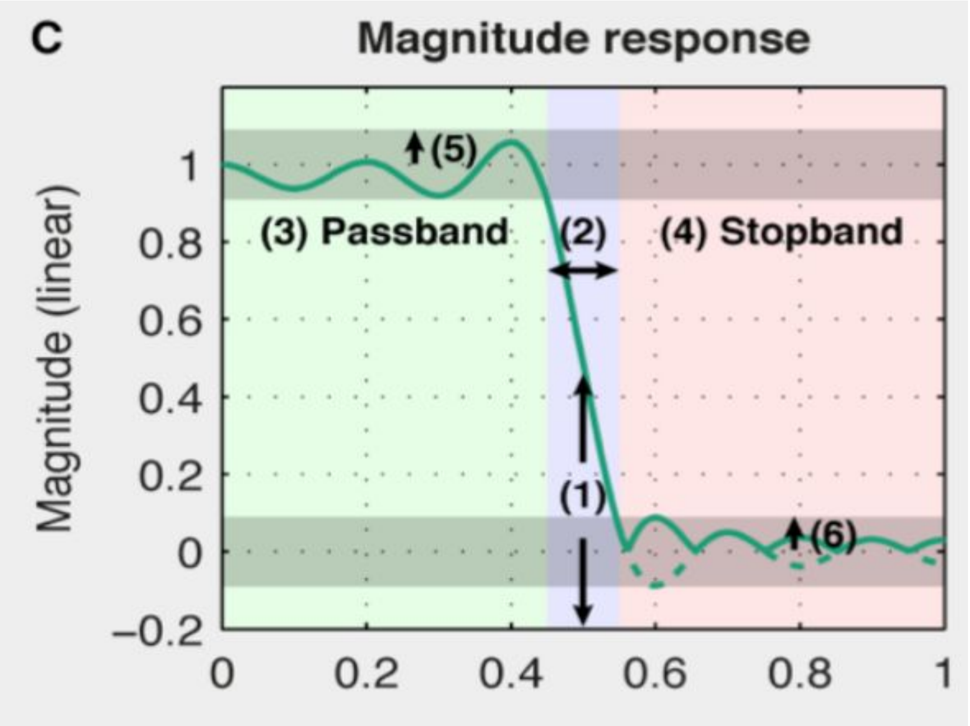
Filter design – Cutoff Frequency

- The **cutoff frequency** separates passband and stopband of the filter and always lies in the transition band



Filter design – Cutoff Frequency

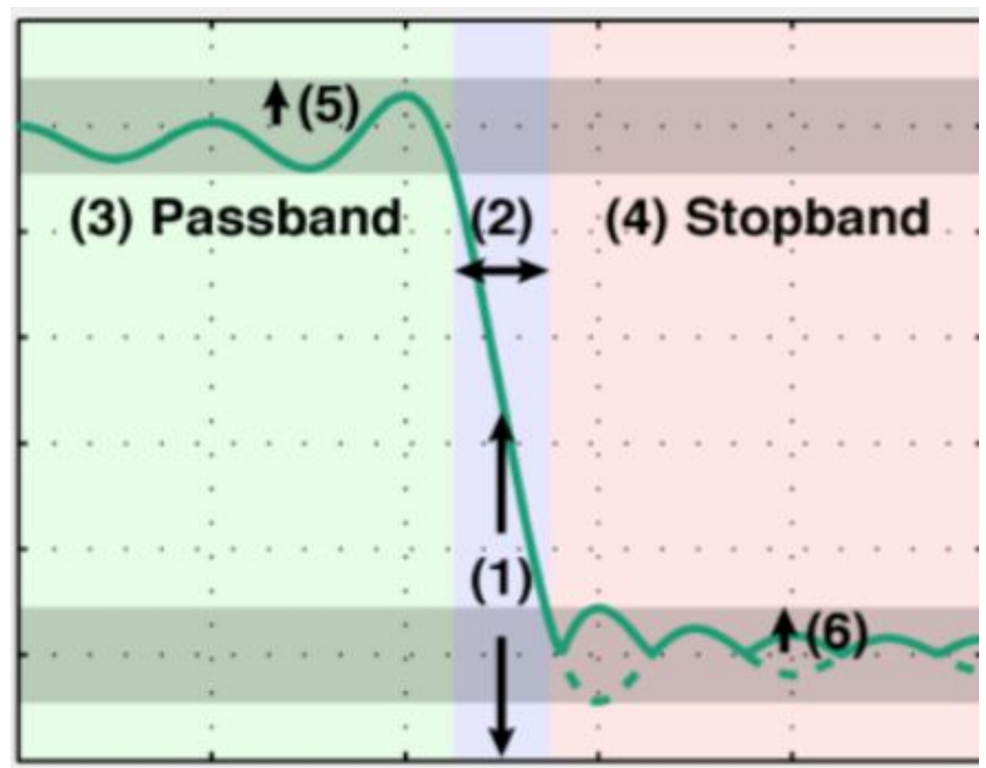
- Different definitions of cutoff frequency are used: **−3 dB (half-energy) cutoff (common for IIR filters) and −6 dB (half amplitude) cutoff (common for FIR).**



Filter design – Roll-off, transition bandwidth, and filter order

Filter design – Roll-off, transition bandwidth, and filter order

- The transition region between passband and stopband enclosing the cutoff frequency is defined as the transition band. **For most FIR filters the −6 dB cutoff frequency is at the center of the transition band**.



Filter design – Roll-off, transition bandwidth, and filter order

- The slope of the magnitude response in the transition band is termed roll-off.
- Filters with a steep roll-off can better separate signal and noise components in adjacent frequency bands than filters with a shallow roll-off.
- **The filter roll-off is a function of the filter order.**

Filter design – Roll-off, transition bandwidth, and filter order

- Longer filters produce stronger signal distortions and also produce a wider temporal smearing of distortions and ringing artifacts.

- Thus, shorter filters with wider transition bands are preferable where possible.

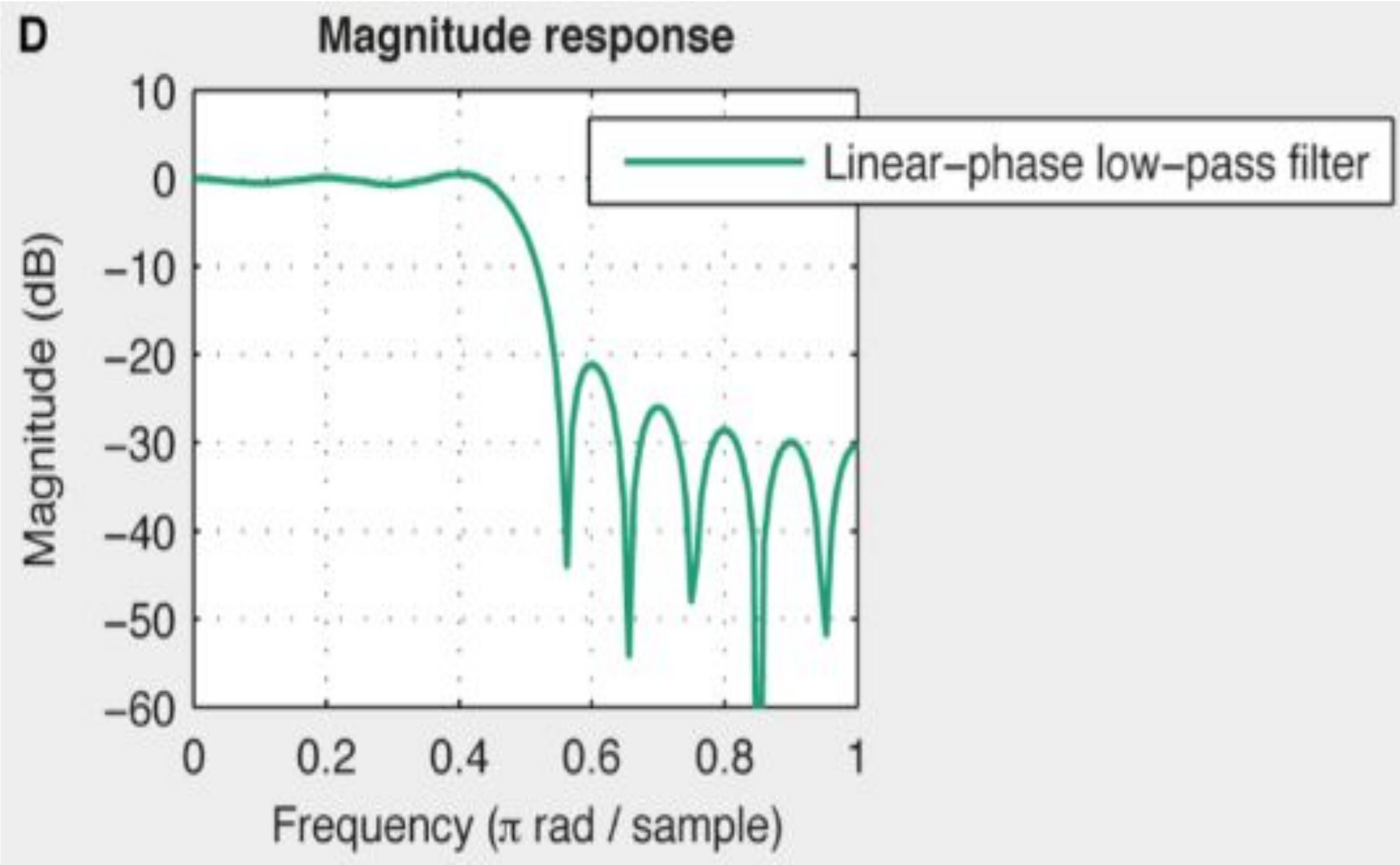
Filter design – Passband ripple/stopband attenuation

Filter design – Passband ripple/stopband attenuation

- The practically achieved magnitude response usually **deviates** from the requested magnitude response (**one in the passband and zero in the stopband**).

- This deviation is commonly termed **passband ripple** in the passband and **stopband attenuation** in the stopband

Filter design – Passband ripple/stopband attenuation



Filter design – Passband ripple/stopband attenuation

- Passband ripple is reported as maximal passband **deviation** in linear or logarithmic units.

- With a passband deviation of, for example, 0.01, the filter output does not amplify or attenuate the signal by more than 1% in the passband (0.086 dB).

Filter design – Passband ripple/stopband attenuation

- Stopband attenuation is reported most commonly in logarithmic units.

- With a stopband attenuation of −60 dB (or 0.001), the signal is **attenuated** by a factor of 1000 in the stopband.

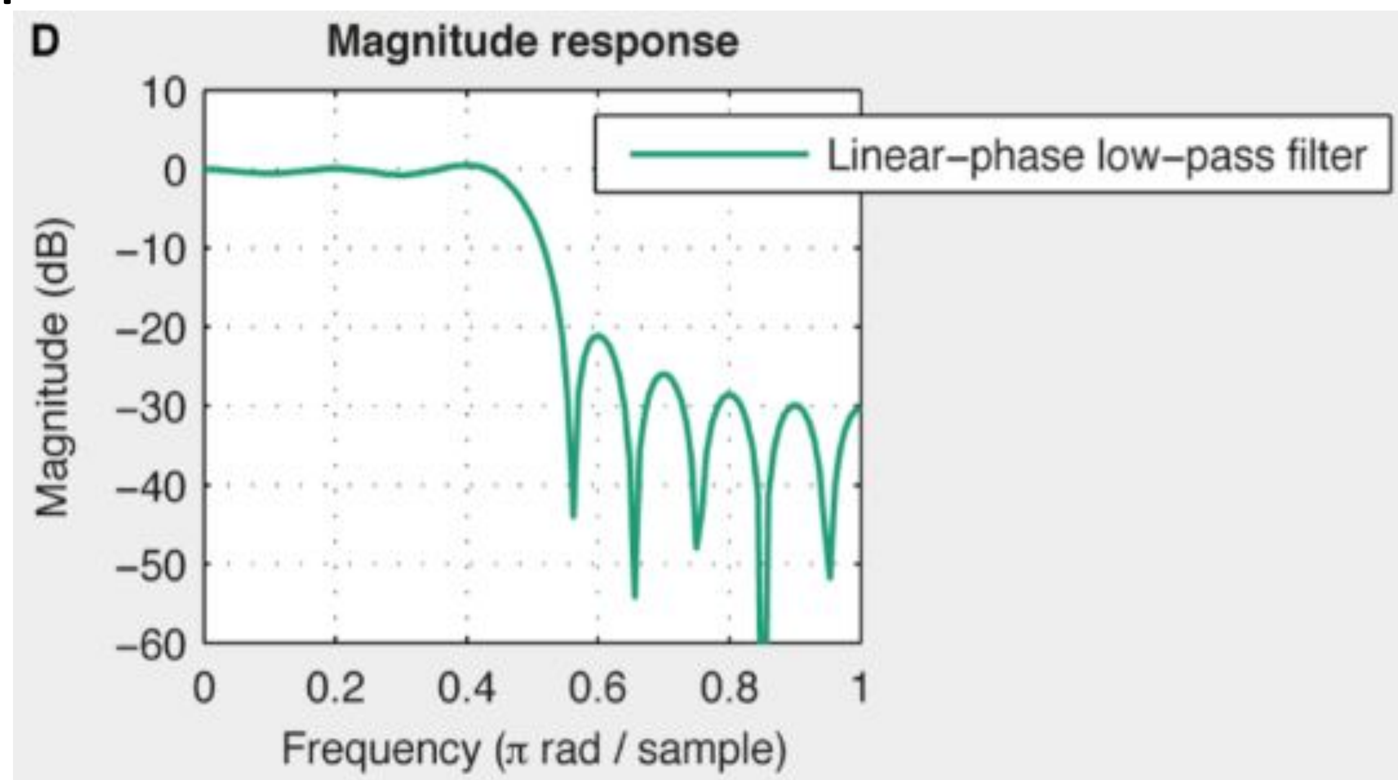
Filter design – Passband ripple/stopband attenuation

- For instance, passband ripple of 0.002–0.001 (0.2%–0.1%) and −54 to −60 dB stopband attenuation are reasonable values for biosignal applications.

- For high amplitude low-frequency noise (near DC), a stopband attenuation of −100 dB or stronger might be required.

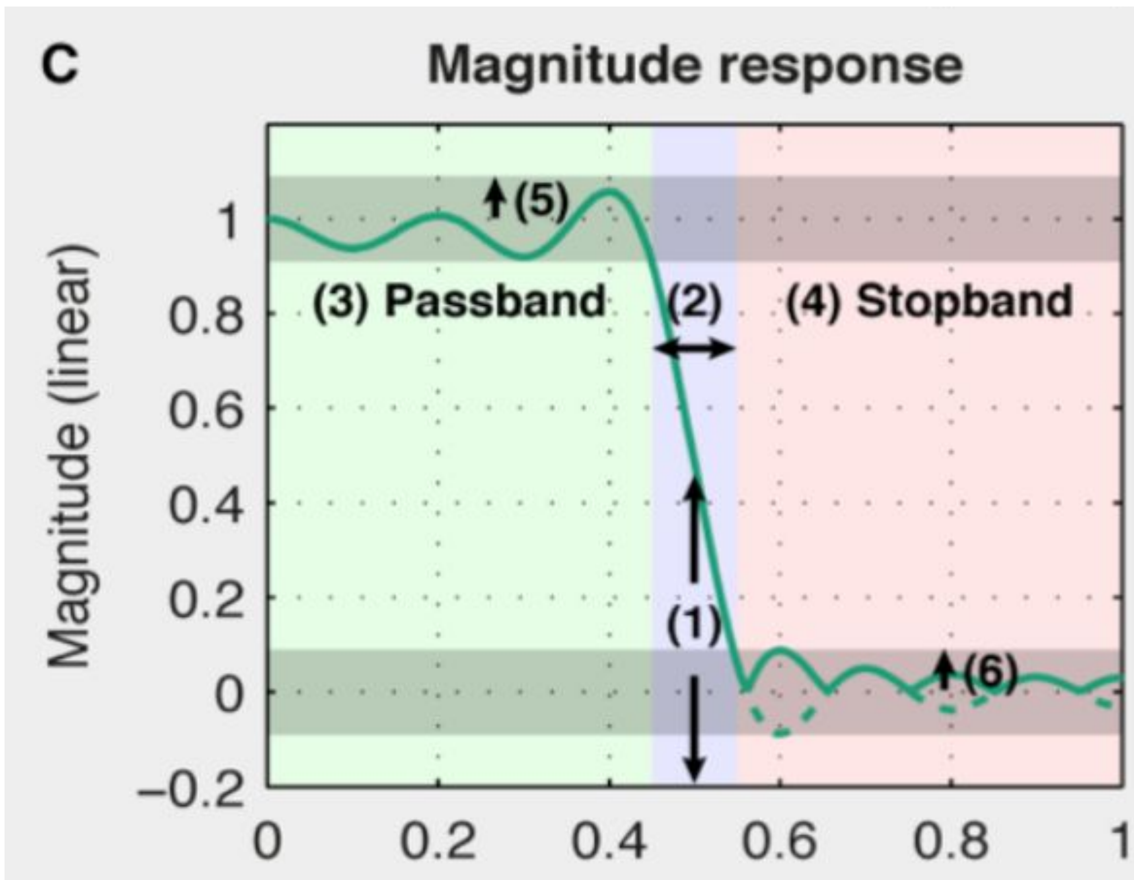
Filter design – Passband ripple/stopband attenuation

- Stopband ripple/attenuation is best evaluated in the logarithmically scaled magnitude response



Filter design – Passband ripple/stopband attenuation

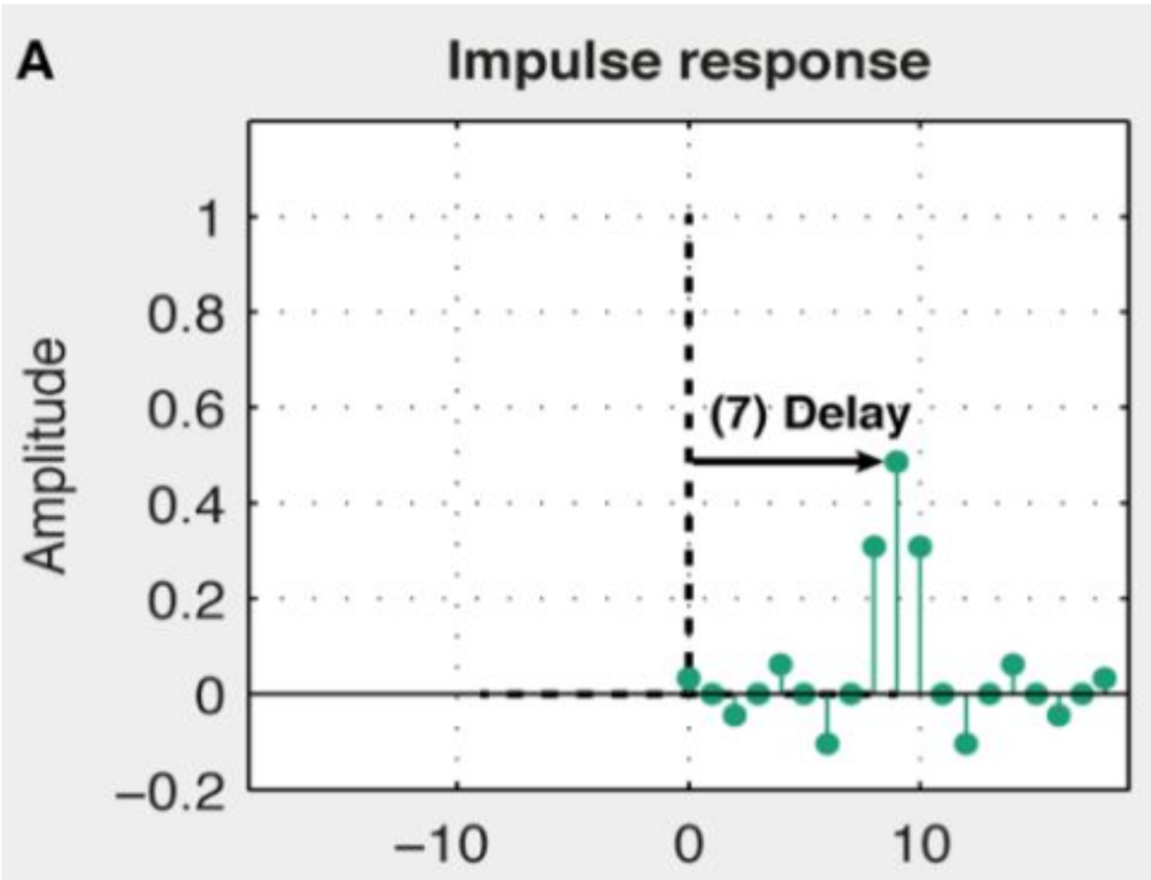
- Passband ripple is better evaluated in the linearly scaled magnitude response



Filter design – Delay

Filter design – Delay

- Every (non-trivial) filter necessarily delays the filter output relative to the filter input



Filter design – Delay

- The most relevant parameter for electrophysiological applications is the **group delay**, defined as the delay of the **envelope** of the signal at a particular frequency.

Filter design – Delay

- Linear-phase filters introduce an equal (group) delay at all frequency bands – the slope of the phase response is constant within the passband.

- Consequently, a signal with all its spectral components in the passband **will not change its temporal shape**.

Filter design – Delay

- The group delay of linear-phase filters can be easily computed based on the length of the filter's impulse response as $(N - 1) / 2$ (in samples).

To do …

conda

If you use conda/Anaconda environments, librosa can be installed from the *conda-forge* channel:

```
conda install -c conda-forge librosa
```