<https://www.frontiersin.org/articles/10.3389/fpsyg.2012.00233/full#FA1>

<https://home.uni-leipzig.de/~biocog/eprints/widmann_a2015jneuroscimeth250_34.pdf>

The low signal-to-noise ratio of EEG and MEG recordings makes filtering an indispensable tool for the analysis of electrophysiological data. However, filtering is also prone to introducing severe distortions in-to the data, biasing or even invalidating the results (see, e.g., VanRullen, 2011; Acunzo et al., 2012; Zoe-fel and Heil, 2013 for prominent examples and Luck, 2005 for discussion).

For many ERP/F applications, in particular unguided, exploratory ERP/F analysis, it is recommended to re-frain from high-pass filtering or to apply very low (≤0.1Hz) cutoff high-pass filters (Acunzo et al., 2012; Luck, 2005). The signal-to-noise ratio in unfiltered electrophysio-logical recordings might, however, be too low for a specific data analysis. Filtering can be a recommend-able option in this case–however, the authors should verify that filtering actually improved the signal-to-noise ratio in the data analysis. However, high-pass filtering replacing baseline and drift correction can be a valid means to remove strong low-frequency (near DC) interferences, e.g., in ERP/F analysis of ongoing speech. Even higher high-pass cutoff frequencies accepting a minor attenuation of evoked components such as the N400 can be reasonable (see e.g., Maess et al., 2006).

Filteringthe data with a 0.75Hz high-pass filter (Fig. 7) has only minor effects on the standard waveform because this carries only moderate energy at low frequencies. The deviant waveforms, however, show significant distortion with a reduced P3 peak amplitude non-causally carrying over the distortion into pre-ceding components, resulting in artificially enhanced N1/MMN and N2 peak amplitudes (relative to pre-stimulus baseline but not relative to neighboring peaks and troughs; see, Acunzo et al., 2012 for detailed discussion). Rousselet demonstrated that causal filtering is not feasible for low-pass filters due to the introduced (non-linear phase) delay, but might be an option for high-pass filtering. Indeed, causally filtering the example data with a 0.75Hz minimum-phase converted filter reveals no indication of artificially enhanced N1/MMN and N2 components (rather, peak amplitudes are slightly reduced due to the attenuation of low-frequency components. Due to the non-linear phase characteristic, the minimum phase filter, however, dramatically distorts the temporal dy-namics of theP3 component. Thus, a causal non-linear filter can be employed to separate long-and short-latency components, avoiding non-causal effects, but should be used carefully because it introduces a de-lay and signal shape distortions.

**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. Linear-phase filters have a perfectly **symmetric** impulse response (or antisymmetric only changing sign between left and right half). 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). **non-linear-phase filters** with an **asymmetric** impulse response introduce different delays in different frequency bands (see Fig. 2 for examples). Thus, non-linear-phase filters distort the temporal shape of spectrally complex or broadband signals (such as ERP components) even if all spectral components are in the passband (and they disturb cross-frequency phase relationships if analyzing phase-phase or phase-amplitude coupling in time-frequency analysis).

**The delay of linear-phase filters can be corrected by shifting the filter output back in time, resulting in a zero-phase filter having no delay** (see Fig. 2). Due to the shift, each sample in the filtered output signal is computed from preceding (past) and following (future) samples of the unfiltered input signal; the filter is therefore classified as **non-causal**. In practice this means that the signal in the smoothed zero-phase filter output **might already deviate from baseline before signal onset in the input, possibly systematically underestimating onset latencies after low-pass filtering** (cf. the step responses in Fig. 2B; see Rousselet, 2012; VanRullen, 2011; Widmann and Schröger, 2012 for discussion), introducing non-causally smeared artificial or artificially enhanced components after high-pass filtering (Fig 7D; see Acunzo et al., 2012 for discussion), or smearing post-stimulus oscillations into the pre-stimulus interval leading to spurious interpretations of pre-stimulus phase (Zoefel and Heil, 2013).

A causal filter, in contrast, computes the output only on the basis of preceding (past) input samples. The step response of a causal filter does not exhibit signal changes due to the step (for example smoothing or ringing) before the onset of the step in the filter input (blue line in Fig. 2B and 2G). Importantly,zero-phase (non-causal) filters preserve peak latencies, while causal filters necessarily shift the signal in time. If a causal filter is needed, a non-linear minimum-phase filter should be considered as it introduces only the minimum possible delay at each frequency for a given magnitude response but distorting broadband or complex signals due to non-linearity (see Fig. 2). Causal high-pass minimum-phase (and other non-linear) filters introduce rather small delays (Fig. 2F and G) while causal low-pass (and band-pass and band-stop) filters introduce larger delays even with minimum-phase property (Fig. 2A and B), which is why they are not recommended in electrophysiology (Rousselet, 2012).

Zero-phase delay can also be achieved with non-linear filters by filtering the filter output a second time in the reverse direction (“two-pass filtering”) to compensate for the filter delay (Smith, 1999, p. 331; filtfilt function in MATLAB/Octave). Two-pass forward and reverse filtering results in a **non-causal filter with a symmetric impulse response**. Two-pass filtering (equivalent to concatenating the same filter twice in the two-port model) doubles the filter order and doubles the length of the (effective) impulse response. Thus, the two-pass filter smears the output wider in the time domain. Two-pass filtering squares the magnitude response, which shifts the −3dB half-energy and the −6dB half-amplitude cutoff frequencies, and needs to be reported properly (attenuation at the one-pass −3dB cutoff is enhanced to −6dB for IIR and at the one-pass −6dB cutoff to −12dBfor FIR filters; see Section 2.7below; Edgar et al., 2005). Two-pass filtering enhances (squares) passband ripple and stopband attenuation. Different software implementations use different strategies to compensate for the doubled filter order and shifted cutoff frequencies. For rep-licability it is thus important to report cutoff frequencies together with not only their definition but also whether they apply to one-pass or two-pass filtering including possible adjustments of order and cutoff frequency. Importantly, the shapes of the (squared) magnitude response of a two-pass filter and the equivalent one-pass filter of double the filter order (having the same effective impulse response length) can be significantly different in particular in frequency bands near the cutoff frequency. One-pass linear-phase filters (corrected by shifting) can achieve a similar magnitude response shape (in particular steepness at the cutoff frequency but not stopband attenuation) at lower orders than a corresponding two-pass filter. **This makes linear-phase one-pass filtering preferable in many applications** (see Fig. 4B).

Various practices and recommendations exist for **low-pass** cutoff frequencies in ERP/F analysis including the suggestion not to apply low-pass filters at all (VanRullen, 2011). Indeed, low-pass filters frequently serve primarily cosmetic purposes as high-frequency noise (except line noise) usually has low energy in electrophysiology recordings (cf. e.g., Widmann et al., 2012, who recorded children’s EEG in electrically unshielded rooms at primary schools; 100 Hz low-pass cutoff filtered ERPs only show very moderate noise levels). Furthermore, later steps in data analysis, such as computing mean amplitudes in given time windows, are low-pass filter equivalents. As a rather unspecific recommendation, we suggest applying low-pass filters with cutoff frequencies higher than 40Hz during ERP analysis, thereby preserving the visible high-frequency components in the ERPs such as the sharp peak of the low latency (P1) compo-nent.