

INTERACTIVE TOOL FOR DEMONSTRATION OF THE EFFECT OF CAUSAL AND ACAUSAL FREQUENCY FILTERING ON SEISMIC SIGNALS

Petr KOLÁŘ

Geophysical Institute, Boční II, 141 37 Praha 4 – Spořilov
Corresponding author's e-mail: kolar@ig.cas.cz

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ABSTRACT

Although causal and acausal (zero-phase) frequency filtering are procedures routinely used in seismic signal processing, many users of respective software tools are not fully aware of at first glance subtle, but for the exact interpretation of the filtered time series significant, differences between the two methods. Typical effects are waveform distortions due to the non-linear phase response in the former case, and generation of spurious anticipatory signal components due to the acausality of the process in the latter. In the present paper an interactive MATLAB program for the tutorial demonstration of the mentioned effects is described. The program allows processing of external (real) seismograms as well as internally generated synthetic seismic signals. All parameters specifying the synthetic signal and the filter properties (type, order, corner-frequencies) can be entered interactively. The amplitude and phase responses of the filters and the results of the filtering procedure are displayed, and the kinematic as well as dynamic parameters of the filtered signals such as onset times, time shifts, time differences and amplitudes can be picked with the cursor. Typical examples of causal vs. acausal filtering are given to demonstrate the performance of the program and the differences between the two approaches (there are presented two synthetic examples to illustrate basic filter behaviour and two examples of real seismograms analysis).

KEYWORDS: effect of causal and acausal filtration; seismic signal processing

INTRODUCTION

Modern seismological research is based on detailed analyses of digital seismograms. A basic step necessary to extract aim-oriented useful information from seismograms, especially from broadband records, is frequency filtering. Digital frequency filtering is a mature field in which the effects of different methods are well understood, but it is not a discipline in which all practicing seismologists or students of seismology are well trained. It therefore may still appear that effects caused by the filtering are erroneously interpreted as natural phenomena. A frequent matter of confusion is the difference between causal and acausal filtering¹⁾. In the present paper a MATLAB program for the demonstration of these differences is presented together with few typical examples. The program is an extended version of that previously described by Plešinger and Kolář (2002).

GENERATION OF SYNTHETIC SIGNALS

Synthetic seismic signals are generated by convolution of a source wavelet in the form of a

Berlage function with an impulse train (Green function) simulating reflections or transitions of the source wavelet along its propagation path.

The Berlage signal is defined by

$$y(t) = t^A \exp(Bt) \sin(2\pi ft), \quad (1)$$

where t is time and A , B , f are constants specifying the shape of the signal: A its rise time, B its damping (duration), and f its predominant frequency. The program offers also the generation of more complex source wavelets in the form of the sum of two different Berlage signals shifted in time.

The wavelet is convolved with a Green function consisting of an arbitrary number of Delta functions defined by their amplitude, polarity and time delay, and noise generated as optional running average of a high-frequency random signal can be superimposed to the resulting convolatory signal.

As an alternative input, the program accepts as well external data in the form of equidistantly sampled time series, e.g. real seismograms.

¹⁾ During so called acausal filtration it can appear foul signal preceding the filtered signal itself. This precursor is fully artificial and does not exist in the nature. Note, that causally filtered signal, which does not contain this foul precursors, suffer from other distortions (as shown e.g. below in the text).

FILTER SPECIFICATION AND CONSTRUCTION

The program enables the user interactively to specify and modify the desired type (lowpass, highpass, bandpass, bandstop), the order (N), and the corner frequencies (f_{lower} , f_{upper}) of the filter to be applied to the synthetic or imported signal. The filter order must be even, odd values are rejected. The program then constructs a corresponding Butterworth (maximally flat amplitude response) approximation of the specified properties of order N .

SIMULATION OF CAUSAL AND ACAUSAL FILTERING

Both types of filtering are simulated by passing the signal through the $N/2$ -order filter twice: in the causal case directly, in the acausal case first directly, and the output of the first passage then in reversed time²⁾. In both cases the amplitude response of the whole (N -order) filter is identical but the phase response is not: the phase distortion caused by the causal filter is the double of that of its $N/2$ -order section, whereas that of the acausal filter is zero (the "direct" and "reverse" phase distortions mutually cancel)³⁾.

The restriction of the filter category to even-order Butterworth approximations may be fully sufficient for the intended demonstration and educational purposes. Rigorous methods of the construction and application of causal and acausal filters with arbitrary transfer functions have been described in detail elsewhere, e.g. in the monography by Plešinger et al. (1996), some basic information can be found also in the "MATLAB - Signal Processing Toolbox" help.

PROGRAM CONTROL AND GRAPHICAL OUTPUT OF THE RESULTS

An example of the configuration of windows displayed by the program is shown in Fig. 1. All relevant input parameters (filter type and order, corner frequencies, signal shape, amplitudes and time shifts of Delta functions which compose Green function, noise parameters, etc.) can be entered interactively from the narrow right-hand control panel in Fig. 1a. Particular sets of parameters can be saved in text files and re-entered again if desired.

The following inputs and outputs are plotted: synthetic (or imported) wavelet, impulse train, seismogram (signal convolved with impulse train),

seismogram plus noise - see bottom window in Fig. 1a; amplitude and phase responses of the applied filter - see top window in Fig. 1a and final signals (unfiltered, causally filtered, acausally filtered) - see separated window in Fig. 1b. The signal window can be arbitrarily zoomed in and out, the kinematic and dynamic parameters of the individual seismic phases can be interactively picked using the cursor, and time differences specified in this way are displayed. The effects of interactive changes of optional parameters are displayed immediately so that the user can easily optimise them according to his needs.

EXAMPLES

As the effects of different methods of frequency filtering on seismic signals have been thoroughly explained many times (see, e.g., Plešinger et al., 1996), we present here only two synthetic examples to demonstrate basic signal distortion and two examples of real data processing.

The first synthetic example (Fig. 2) demonstrates the time shift of signal maxima caused by causal filtering, and the second (Fig. 3) the appearance of spurious pre-cursors in acausally filtered signals. These two basic examples indicate that if we are interested in exact values of signal maxima and their time positions, acausal filtering is the preferable method. On the other hand, if we are looking for exact signal onsets, acausal filtering can lead to serious misinterpretations⁴⁾.

In Figure 4 there is an example of analysis of a suspicious long period pulse superposed over a real seismogram of a local earthquake. Onset of the pulse can be detected in filtered signal and its probable origin can be then explained.

The last example (Fig. 5) comes from an active refraction measurement CELEBRATION 2000 (Guterch et al. 2003). Due to build in lowpass antialias acausal filter in used seismometers REFTEK false signals can be seen in the records of explosion, namely for records in small epicentral distances, i.e. for records containing relatively high-frequency signals. This effect must be taken into account when times of first onsets are picked (Hrubcová et al.).

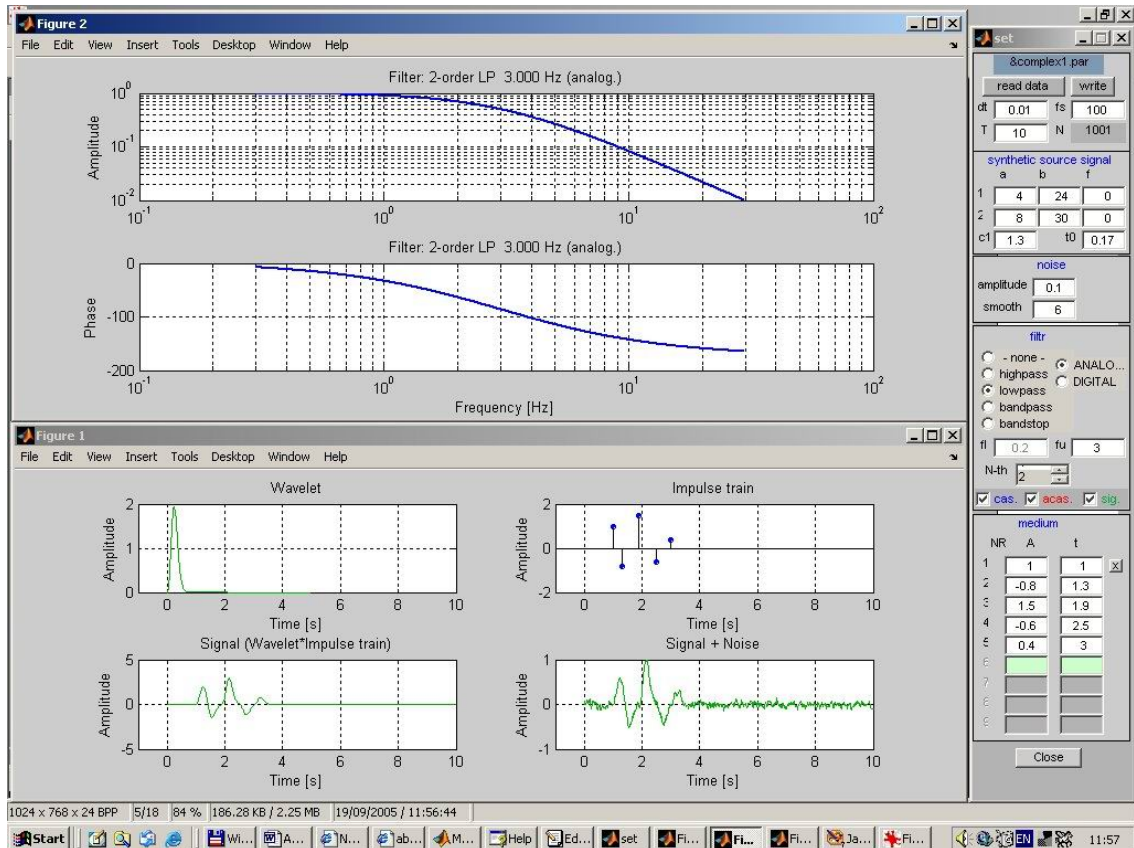
TECHNICAL NOTES – PROGRAM DISTRIBUTION

The described tool is available on request (see the contact in the article header). The author is able to provide either the MATLAB source code (MATLAB

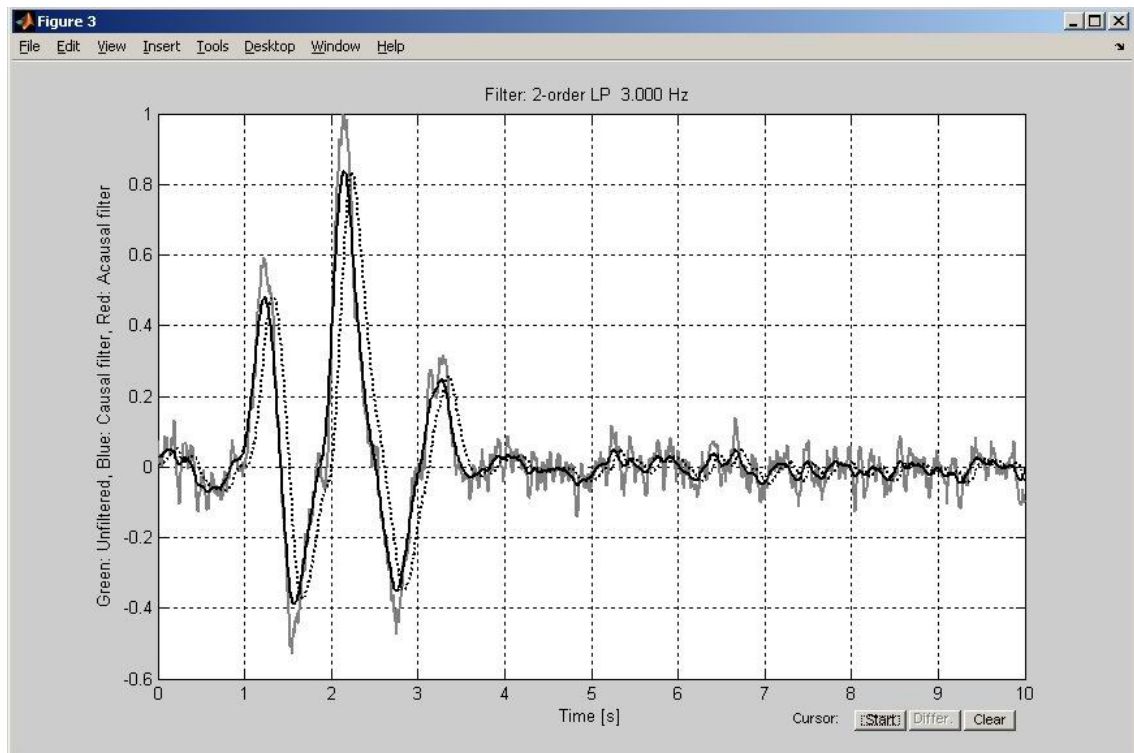
²⁾ Acausal filters are typically realised by recursive formulas (so called Infinite Impulse Filters), causal filters then by convolution formulas (so called Finite Impulse Filters or also zero-phase filters). However this is not the rule and the filters can be realised by other particular algorithms, which is also our case.

³⁾ In the code, the filtration is realised by standard MATLAB functions `filter` and `butter`. Note, that our acausal algorithm is almost identical with function `filtfilt`.

⁴⁾ Note, that filtration is a linear process (in mathematical sense; i.e. if input signal is multiplied by a constant, the output is multiplied by the same value etc.) and all observed signal distortions are consequence entirely of non-constant course of phase characteristic of filter.



a



b

Fig. 1 Screen shot – a typical example of the screen displayed by the interactive tutorial program; for details see section *Program control and graphical output of the results*. Even if the purpose of the figure is to show outlook of graphical outputs, effects of filtration, discussed later in the text, can be recognised already here, namely in Fig. 1b - grey line is nonfiltered noisy signal, black full line is acausally filtered and dotted line is causally filtered signal, filtration with 2nd order 3Hz lowpass filter.

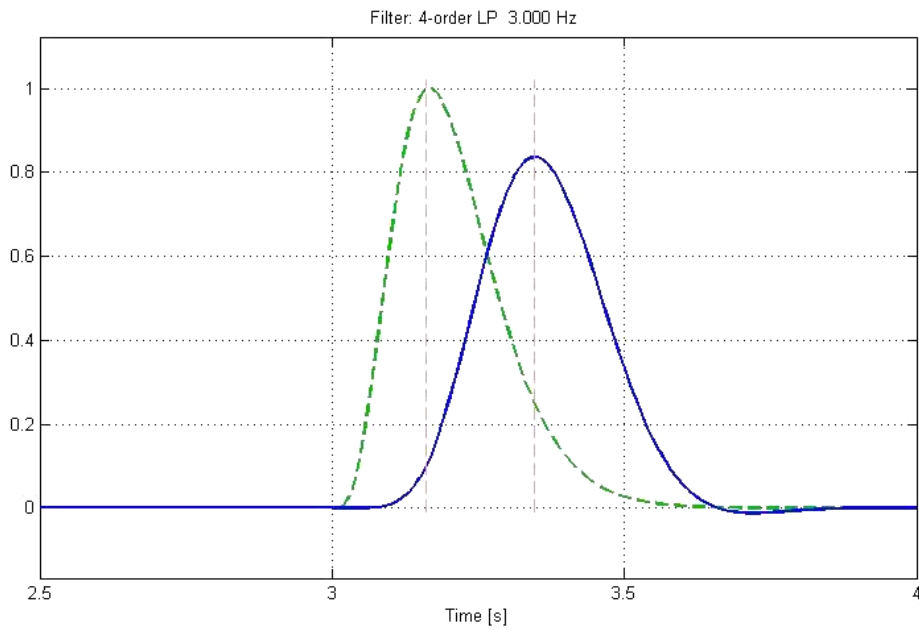


Fig. 2 Example of causal filtration effect: a pulse with sampling interval 0.01s is filtered with a 4th order 3Hz lowpass filter. The dotted line shows the original signal, the full line the filtered one; the two vertical dashed lines indicate the time shift of the positions of the signal maxima (the shift is about 180 ms, which corresponds to 18 samples).

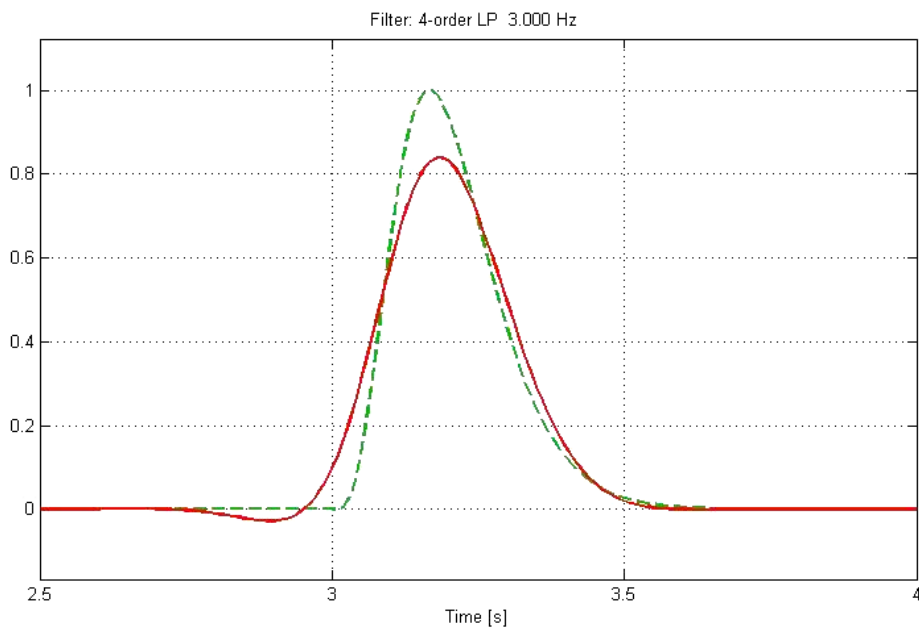


Fig. 3 The same signal and filter as in the Figure 2, but signal (dotted line) filtered acausally (full line). The time shift of the maxima is much smaller, almost negligible, but a false pre-cursor arises 0.3 second before the onset of the unfiltered signal in this case.

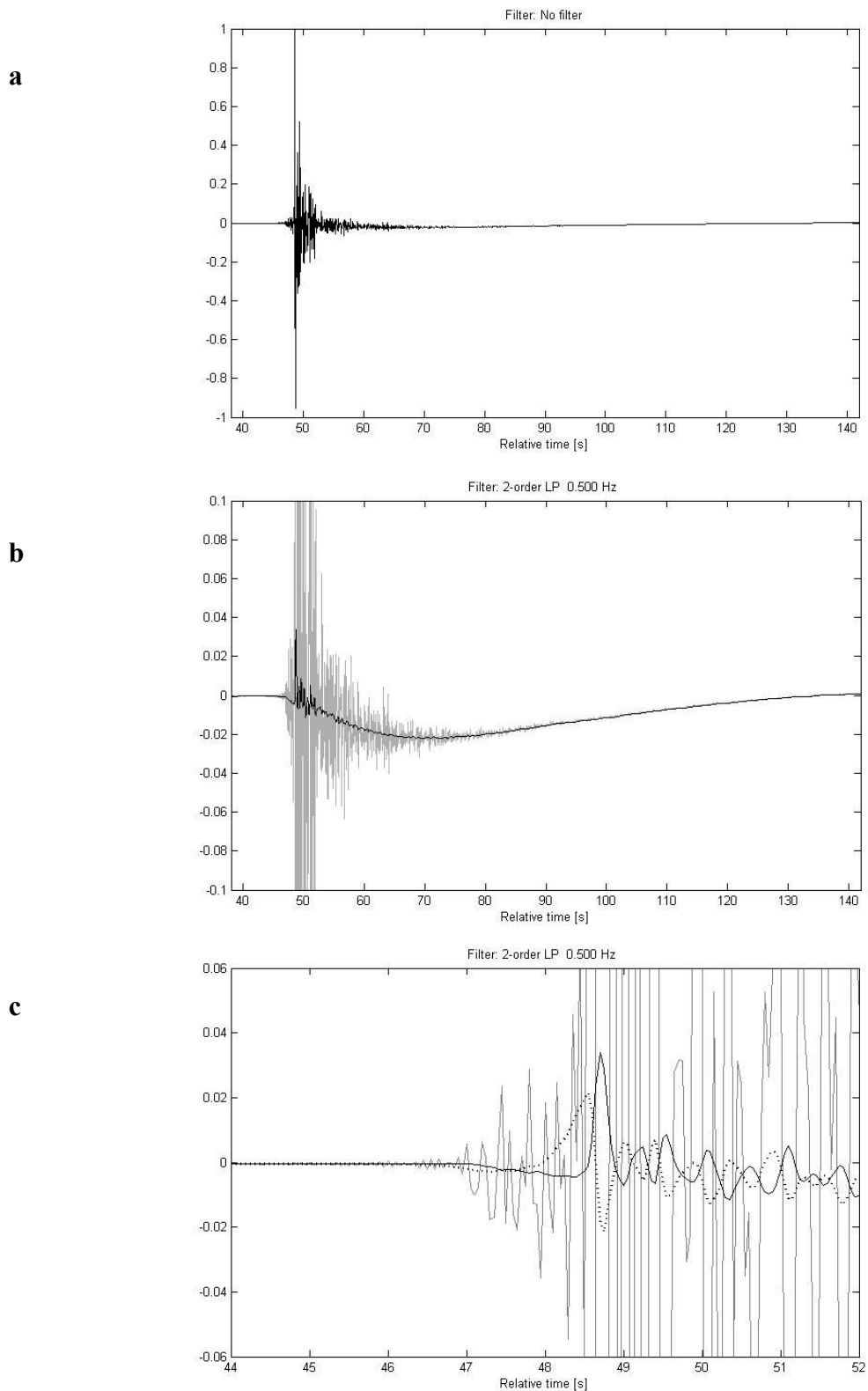


Fig. 4 Record of local earthquake (21:55, Nov, 13, 2002, M_L 3.2, epicentral distance 8km) at station SERG (Sergula, Greece; station is operated by Faculty of Mathematics and Physic, Charles University, Prague). In Fig. 4a there is record of NS component (sampling frequency 20Hz). The long period pulse is hidden in the seismogram and can be hardly seen in this seismogram scale. In Fig. 4b there is the same signal, but zoomed (10x in vertical axes) and filtered with causal lowpass filter 2nd order, corner frequency 0.5Hz (black line) on the background of the original non-filtered signal (grey line). Now the long period pulse (duration about 100s) can be easy seen, as well as its onset joined with S waves arrival can be identified. Zoomed detail (Fig. 4c) shows onset of the pulse and comparison of causal (full black line) and acausal (dotted black line) signal filtration, again on the background of nonfiltered signal (grey line). Note that the filtered signals are almost of opposite phases here. The pulse itself is with high probably generated by local permanently lasting tilt triggered by high-frequency ground vibration – the details can be found in Zahradník and Plešinger (2005), from where the quoted examples is also overtaken.

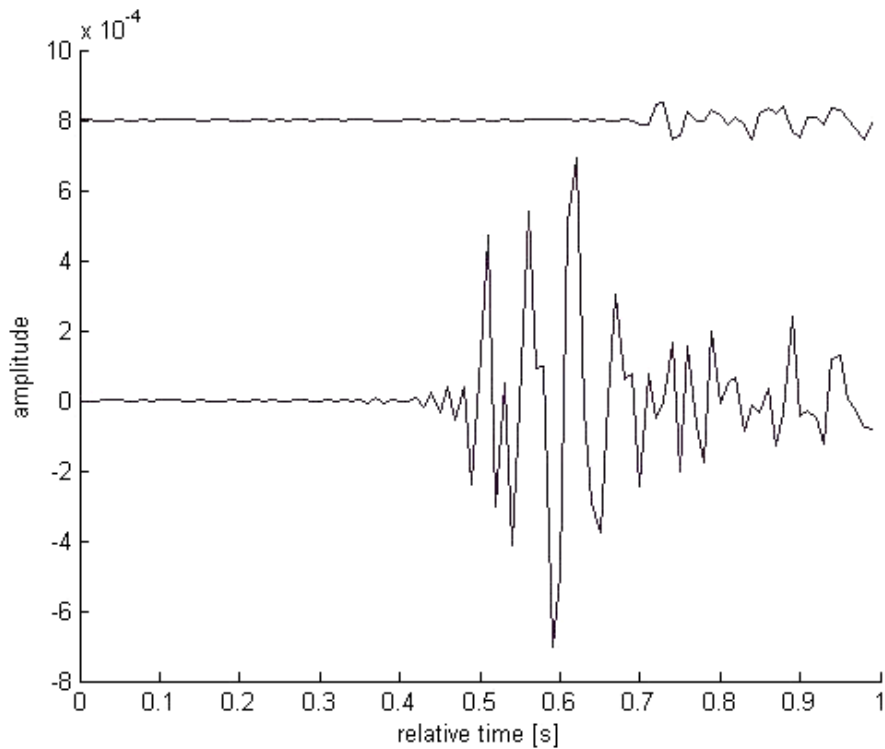
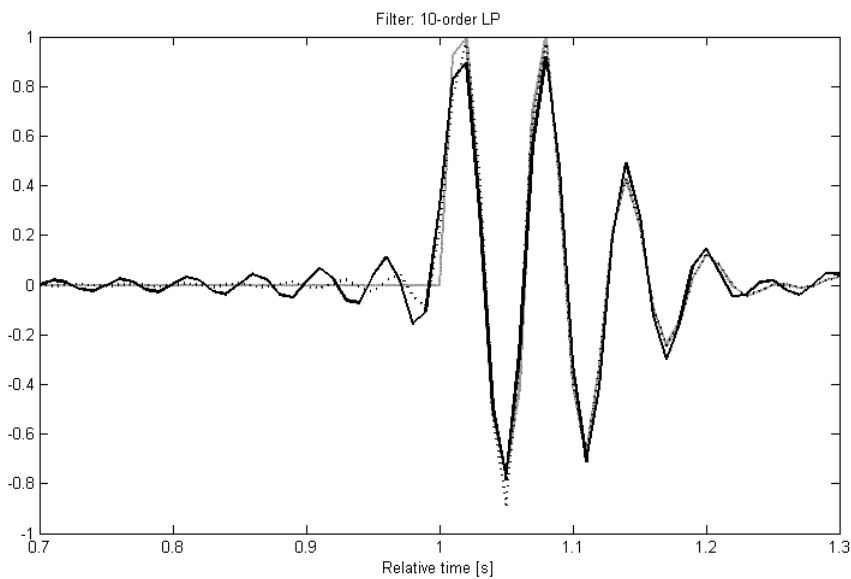
**a****b**

Fig. 5 In Fig. 5a there is an example of records from the experiment (lower seismogram was recorded in distance 10.1 km, the upper one in 15.2 km; sampling frequency is 100Hz, shot number 29050, stations 9120 and 9118 - see Guterch et al., 2003). The decay of amplitudes can be seen as well as simplification of onset of farer signal. In Fig. 5b there is a synthetic simulation of occurrence of the acausal (false) signal. The synthetic testing signal of prevalent frequency 15Hz (grey line) is filtered by steep acausal lowpass filter of corner frequency 20Hz (black full line) and 25Hz (black dotted line). Difference 5Hz corresponds to the observed differences of prevalent frequencies between two seismograms presented in Fig. 5a. The decay of false signal (for time < 1s; t = 1s is onset of synthetic testing signal) is obvious.

incl. Signal Processing Toolbox, version 5.3 or higher is required), or in *.exe version for WINDOWS (Windows98 or higher, inclusive of MATLAB run-time library distributed within the author's MATLAB license). In addition to the program code, the package contains also a short description of the program controls, an installation description, and several basic examples in the form of files of control parameter sets.

CONCLUSION

The exact determination of the basic kinematics and dynamic parameters of different seismic body wave phases such as onset time, onset polarity, rise time, duration, maximum amplitude, waveform complexity etc. from seismograms is a crucial prerequisite for reliable seismological interpretation. Knowledge inferred from these parameters involves such fundamental seismological characteristics as hypocenter co-ordinates, source mechanism, and earth structure along the propagation path. In order to be able reliably to extract the parameters from broadband observations or from recordings of weak seismic events, the analyst must subject the seismograms to frequency filtering prior to the analysis. In order to minimize the effects of this inevitable pre-processing procedure on the waveforms used for the interpretation, the type, order and corner frequencies of the filter and the method of filtering must be chosen very carefully (the applied procedure must be optimised) with respect to the given interpretation purpose, or these effects must be known to the interpreter so that she/he can account for them in the interpretation.

The MATLAB program described in the present short contribution offers the seismologist a simple interactive tool for the tutorial demonstration of the mentioned effects. The included typical examples of signal distortion due to the filtration may serve as starting point for analysts-beginners.

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