

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

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Abstract

Causal (IIR) and acausal (zero-phase) frequency filtering are procedures routinely used in the processing of seismic and other non-stationary signals but many users of respective software tools are not aware of the apparently subtle, for the exact interpretation of the filtered signals however significant differences between the two methods. Typical effects are time-shifts and waveform distortions due to the non-linear phase response in the former case, and generation of spurious artifacts prior to impulse arrivals due to the acausality of the filtering process in the latter. Motivated by teaching experience from training courses on modern methods of seismological data processing and with students of geophysics, we wrote a tutorial MATLAB program for the interactive demonstration of the mentioned effects. The program enables the user to model convolutive input signals, to add noise to them, to choose the type (high-pass, low-pass, band-pass, band-stop), order and corner-frequencies of the filter, and to select the desired method of filtering (causal, acausal, both). The filtered signals are displayed along with the frequency responses of the applied filter, and kinematics and dynamic signal parameters such as onset times, phase shifts, amplitudes etc. can be picked with the cursor. Although the program was developed in particular as educational tool for students in geosciences and for training of observatory seismologists, it may prove useful as well to any non-specialist in digital signal processing who uses frequency filtering as a part of data analysis and interpretation in his field of research.

Introduction

Seismology is a data based geoscientific discipline, and modern seismological studies are based on detailed analyses and interpretations of large sets of digital, mostly broadband seismograms. Before any analysis takes place, the seismograms are almost always subjected to aim-oriented digital processing or pre-processing. Digital signal processing is a mature field and many sophisticated seismological data processing software packages have been developed and are in use worldwide, but not all practicing seismologists are fully familiar with the principles and properties of the procedures these packages offer. Under such circumstances their application holds the potential for seismological misinterpretations due to effects of the data processing procedure unknown to the user.

One of the fundamental pre-processing procedures necessary for proper analysis and interpretation of broad-band seismograms or of noisy recordings of weak seismic events, is causal (IIR) or acausal (zero-phase) frequency filtering. The differences between the two alternatives¹ are well-known among specialists in signal processing but less familiar to seismologists. The effect of the non-linear phase response of an IIR filter within its passband is to produce much larger waveform distortions than a zero-phase filter with the same amplitude response and, in the case of low-pass or band-pass IIR filtering, to cause time

¹ A good frequency selectivity with a comparatively low number of filter coefficients can be achieved with IIR filters. However, IIR filters always cause phase distortions within their passband, and high-order IIR filters with poles near the unit circle can become unstable due to quantization errors. FIR filters are easier to design at the price of reduced flexibility in the amplitude response and can have an exactly linear phase response (constant time shift) which can easily be accounted for in the filtering process, i.e., FIR filters can be implemented as zero-phase filters. FIR filters are insensitive to quantization errors and, therefore, always stable.

delays of impulsive arrivals. Zero-phase filters generate, on the contrary, artifacts prior to impulsive arrivals, which can easily be misinterpreted as seismic signals. Essentially all modern seismic data acquisition systems use zero-phase FIR filters for anti-aliasing, and their effect must be compensated for (changed into a minimum phase characteristic) prior to any analysis of the onset properties of seismic signals (onset times and polarities, analysis of so-called nucleation phases and of slow initial phases).

The aim of the MATLAB program described below is to provide seismologists, students in geophysics and workers in all fields of data-based research in general, a simple, easy to handle tutorial tool for the interactive demonstration of the properties and effects of causal and acausal frequency filtering on non-stationary time series.

Seismogram Simulation

Seismic waves generated by earthquakes belong to the most complex signals observed on our planet. Those of them propagating through the interior of the Earth, so-called body waves, are described by the convolution of the source function (body-wave signal radiated by the earthquake) with the transfer function of the medium through which the source signal propagated to the point of observation. Both functions, especially the latter one, called Green's function, are complex functions of many variables.

For the present purpose the use of essentially simplified source and propagation path functions is not only quite sufficient but even desirable. A convenient source function, allowing modeling of the rise time, the predominant frequency and the spectral selectivity (damping) of the source wavelet of a seismic body wave, is the so-called Berlage function,

$$w(t) = At^\alpha e^{-\beta t} \sin(2\pi ft).$$

The required amplitude, rise time, damping and predominant frequency of the wavelet is modelled by proper selection of the constants A , α , β and f , respectively. For the simulation of more complex wavelets the program offers the possibility of superposing two different, mutually arbitrarily delayed Berlage signals.

The effect of the propagation path is simulated by convolution of the source signal with a Green function in the form of an impulse trace, $i(t)$, consisting of an optional number of Delta functions, $\delta(t_k) = 1$ for $t_k = 0$ and $\delta(t_k) = 0$ for $t_k \neq 0$, each defined by its amplitude, polarity, and time delay,

$$i(t) = \sum_{k=1}^K C_k \delta(t_k), \quad t_k = t - \tau_k.$$

This trivial Green function corresponds to the effect of simple reflections of the seismic wavelet on discontinuities at different depths.

A further option is the superposition of random noise, $n(t)$, to the resulting convolatory signal. Optional parameters are the signal-to-noise ratio and the number of samples over which the random noise is averaged in order to limit its high-frequency spectral components. The final simulated signal thus is defined as

$$y(t) = w(t) * i(t) + \bar{n}(t),$$

where $\bar{n}(t)$ is the filtered random noise.

Causal and Acausal Frequency Filtering

Causal digital filters are constructed using the standard MATLAB function `butter` (Butterworth filter). Interactively entered options and optional parameters are filter type (low-pass, high-pass, band-pass, band-stop), corner frequencies, and order of the filter. Causal filtering is performed by the direct application of MATLAB's `filter` function.

Acausal filtering is implemented by repeated application of the `filter` function using the filter coefficients computed by `butter`, first directly to the original seismogram, then

to the time-reversed result of the direct filtering. The actual order of the acausal filter is doubled in this way.

To preserve the direct compatibility of the results of both methods of filtering, the causal procedure is performed twice as well. Even if this simplified approach restricts the demonstration capacity of the program to the comparison of the effects of causal and acausal filtering with filters of even order with multiple (double) zeros/poles, it fully suffices for the intended 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 special monography by *Plešinger et al. (1996)*.

Program control and graphical representation of the results

An example of the window configuration displayed by the program is shown in Fig. 1. All important input parameters (those of the composite basic wavelet $w(t)$ and of the impulse trace $i(t)$; relative amplitude and smoothing window of the random noise $n(t)$; filter type, order and corner frequencies; method of filtering) can be entered interactively from the control panel (right-hand window in Fig. 1).

The effects of changes of the optional parameters are displayed immediately so that the user can easily optimize them according to his needs. The following characteristics are plotted currently: basic wavelet (source signal), impulse trace (Green's function), noise-free seismogram (source signal * impulse trace), noisy seismogram (bottom window in Fig. 1); amplitude and phase responses of the applied filter (not shown in Fig.1); and output signals (unfiltered, causally filtered and acausally filtered seismogram, top window in Fig. 1).

The signal window can be arbitrarily zoomed in and out, the kinematics and dynamic parameters of the individual seismic phases can be interactively picked using the cursor, and time differences specified in this way are displayed.

Frequently used or default parameter sets can be saved in standard input parameter files and re-entered in the program if desired.

Conclusion

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 the crucial pre-requisite 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 optimized) with respect to the given interpretation purpose, or these effects must be known to the interpreter so that he can account for them in the interpretation.

The MATLAB program described in the present short contribution offers the seismologist a simple but effective tool for this purpose. The motivation for our decision to present it to the broader professional public is our believe that it may prove useful also to workers in other fields of research based on the interpretation of frequency pre-filtered non-stationary time series.

References

Plešinger, A., Zmeškal M., and J. Zedník, Automated Preprocessing of Digital Seismograms: Principles and Software, Version 2.2., E. Bergman, ed., Prague & Golden, 1996.

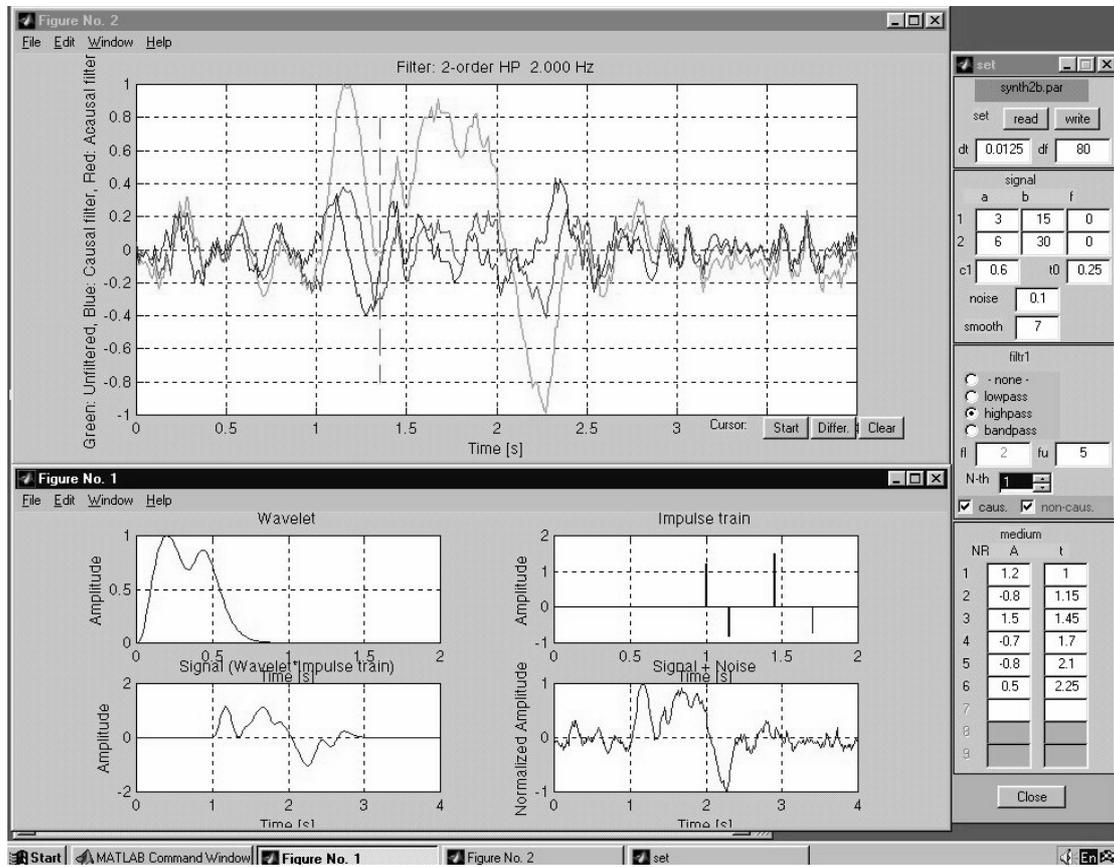


Figure 1. Typical example of the screen displayed by the interactive tutorial program; for details see section *Program control and graphical representation of the results*.