High-Frequency Acausal Arrivals Caused By Non-Causal FIR Filters

These precursors **obscure the initial pulse polarities and first-arrival times**. Is it possible to correct for the acausality? If not, how can we deal with such arrivals?

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Abstract

For the past 20 years, digital recording instruments have oversampled analog waveforms and used zero-phase FIR filters to provide anti-aliasing filtering prior to decimation. For signals with high-frequency energy near the final Nyquist frequency — such as the recordings from stations within a few kilometer epicentral distance of the aftershocks of the 2011 Virginia earthquake — at stations with RefTek RT130 DAS units, the waveforms for very impulsive first arrivals typically had a high-frequency precursor immediately preceding the impulsive first arrival. These precursors obscure the initial pulse polarities and first-arrival times. Fowler (1992) suggested a technique to replace, after recording, the zero-phase FIR filters with a causal filter produced by calculating the causal minimum phase for a filter with the same amplitude response as the FIR filter using properties of analytical functions. That phase delay is a multiple of the Hilbert transform of the log of the FIR amplitude spectrum, and Fowler followed the procedure of Claerbout (1985) that uses the FFT to calculate the Hilbert Transform. Chapman, et al. (1988) used the same procedure to find the causal minimum-phase response for the VTSO SE network responses, but they used a technique that did not in-

volve the FFT. In this report, we compare those two procedures with each other and with a simple 3-pole, low-pass, causal, Butterworth filter that has a corner at 80

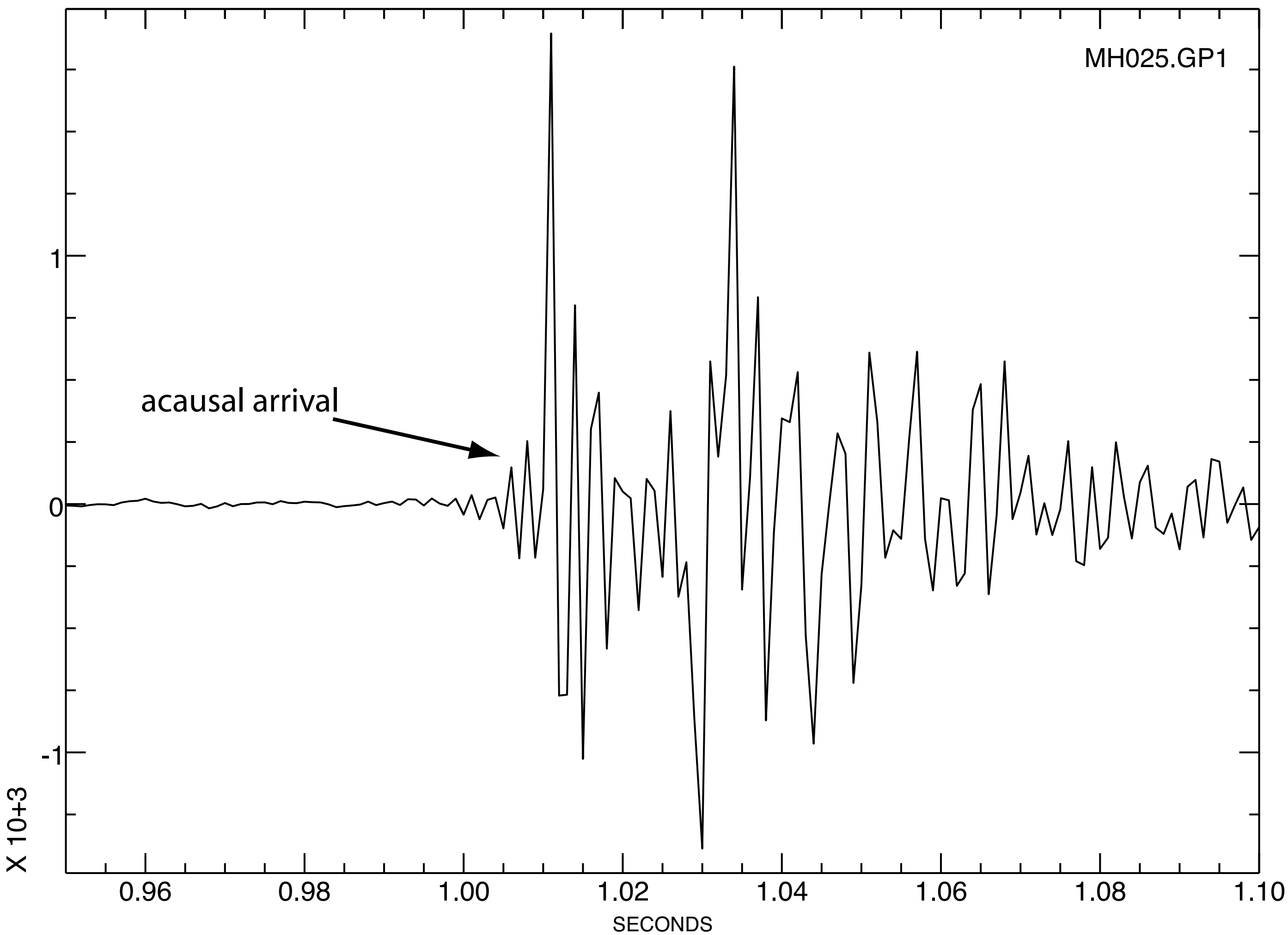
Outline

- Examples of waveforms with the acausal arrivals;
- Discuss and apply a procedure to replace the acausal filter with a causal one:
  - For analytic minimum-phase functions, the phase can be calculated from the amplitude using a Hilbert Transform
- We show that a simple low-pass filter produces comparable results to the HT procedures.

Outline

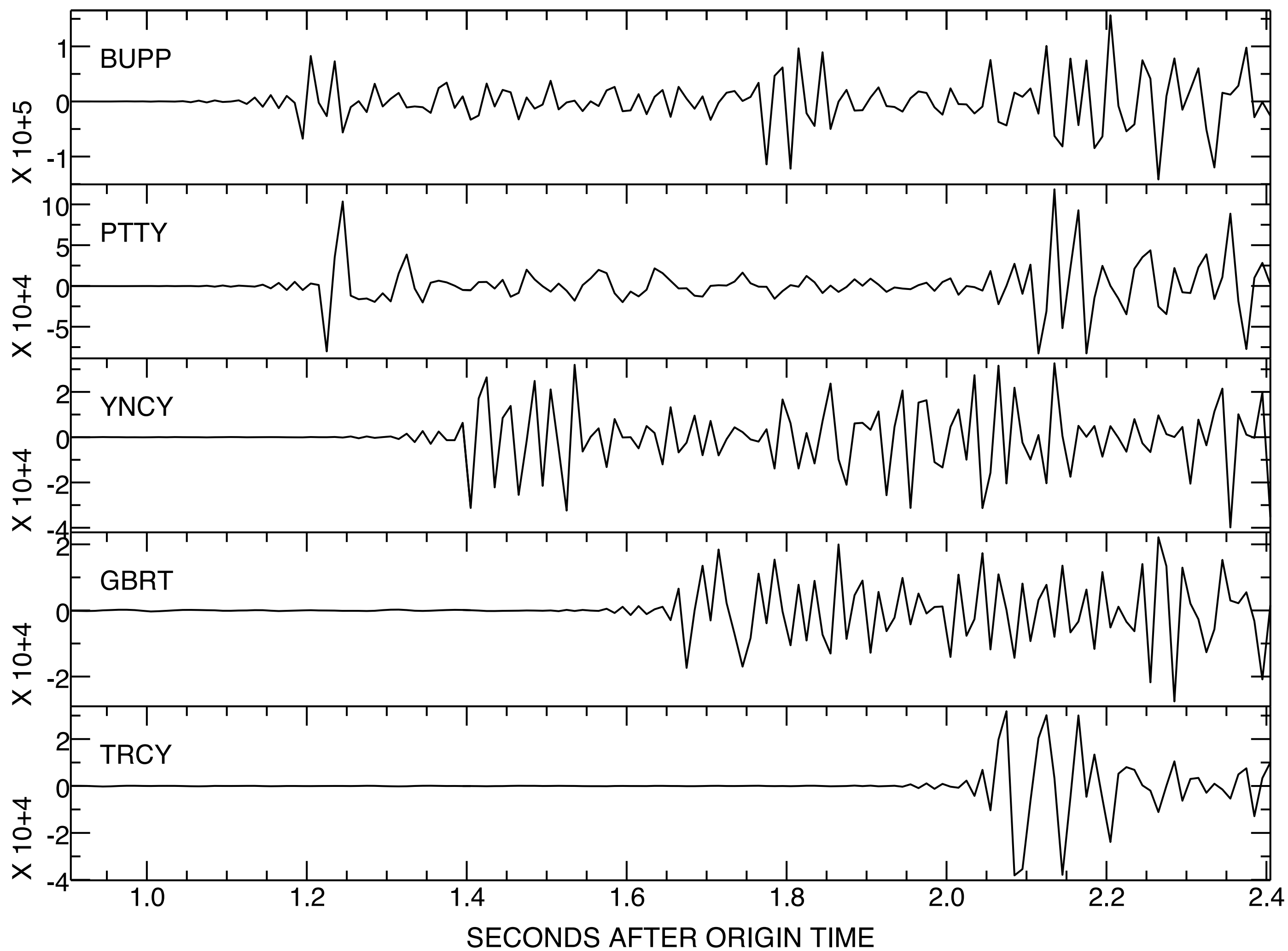
- Examples of waveforms with the acausal arrivals;
- Discuss and apply a procedure to replace the acausal filter with a causal one;
  - For analytic minimum-phase functions, the phase can be calculated from the amplitude using a Hilbert Transform.
  - We introduce two methods to find the Hilbert Transform: one uses the discrete FFT, one uses numerical integration
  - We test these methods on a function for which we know both the amplitude and phase.
  - We apply these to real data.
- We show that a simple low-pass filter produces comparable results to the HT procedures.

Acausal Arrival for a Microearthquake



M_W = -3.2   1000 sps on a RefTek 130 in the SAFOD main hole.

VA Earthquake Aftershock (2011/08/30)



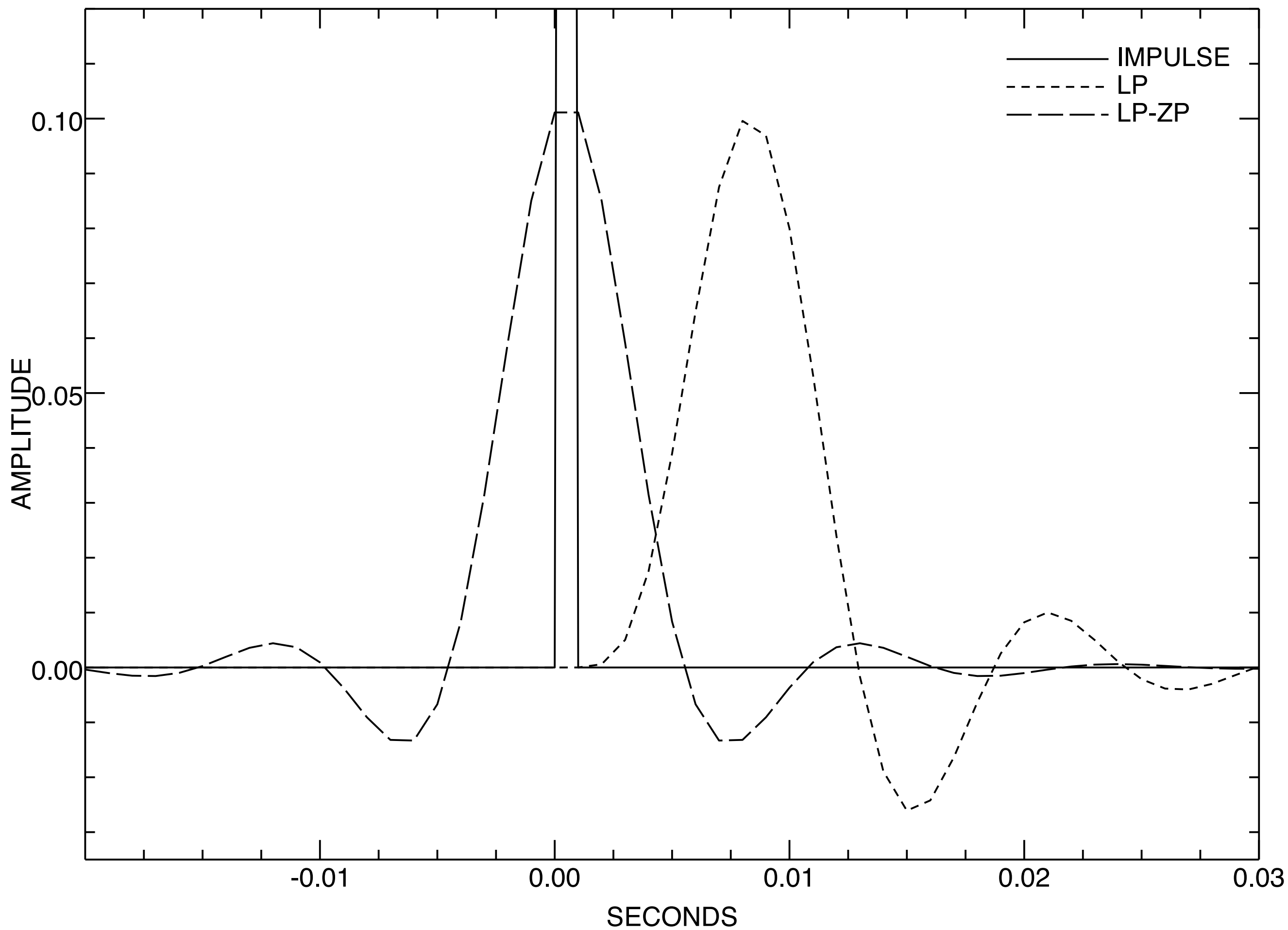
M = 2.0   100 sps on RefTek 130s at VA Tech stations.

Zero-Phase FIR Filter $\Rightarrow$ Causal Filter with Same Amplitude

- For a 6-pole Butterworth filter, we know both the amplitude and phase
- We calculate both the causal and zero-phase impulse response for this filter.
- We calculate the minimum phase from the amplitude of the zero-phase amplitude response using the Hilbert Transform by two methods:

    1. Digital Fast-Fourier Transform (not using SAC)
    2. the numerical method developed by Bolduc (1972)

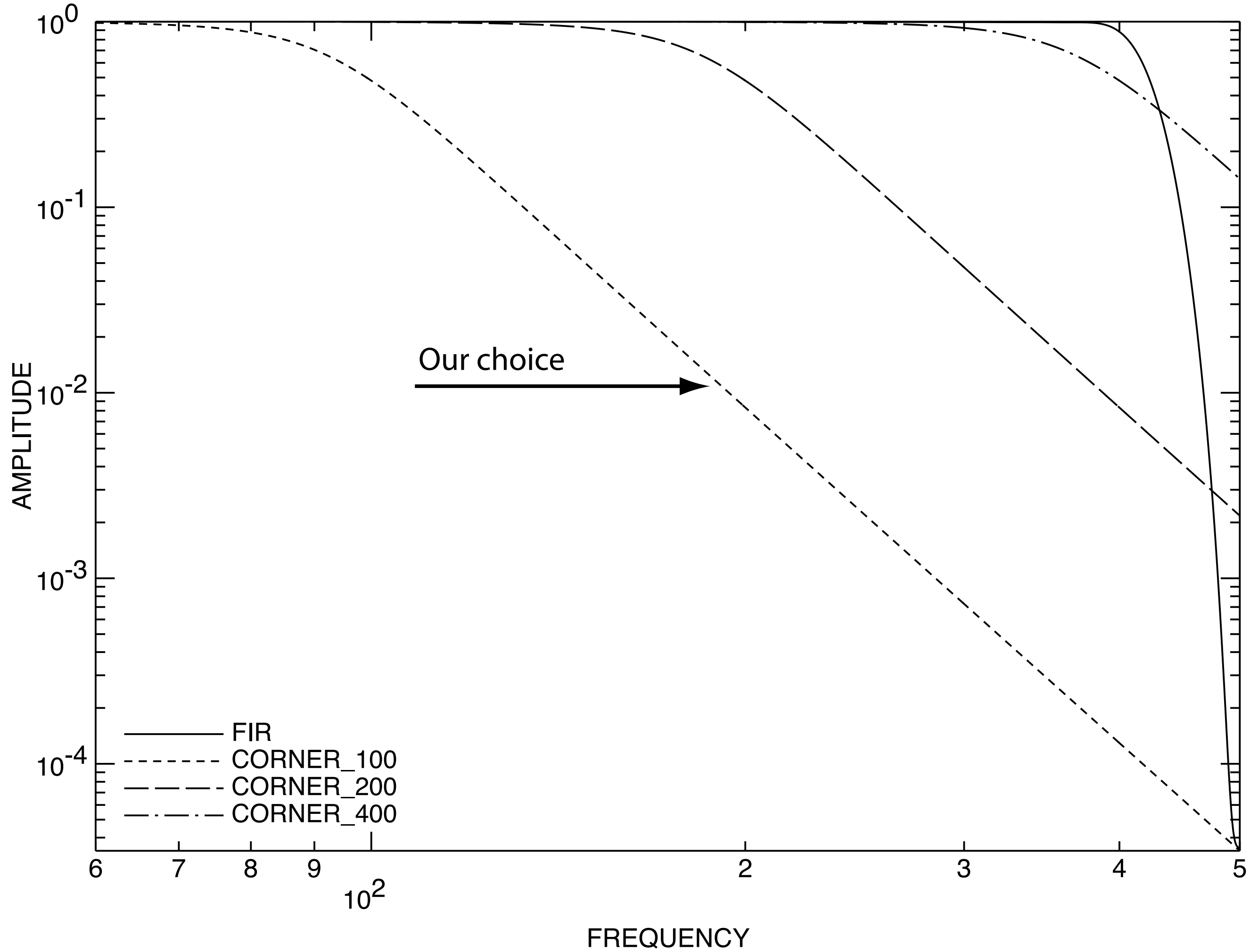
Impulse and Low-Pass Filtered Waveforms



LP-ZP: zero-phase 6-pole with corner at 100 Hz; LP: causal.

Objective: Calculate LP from LP-ZP using HT.

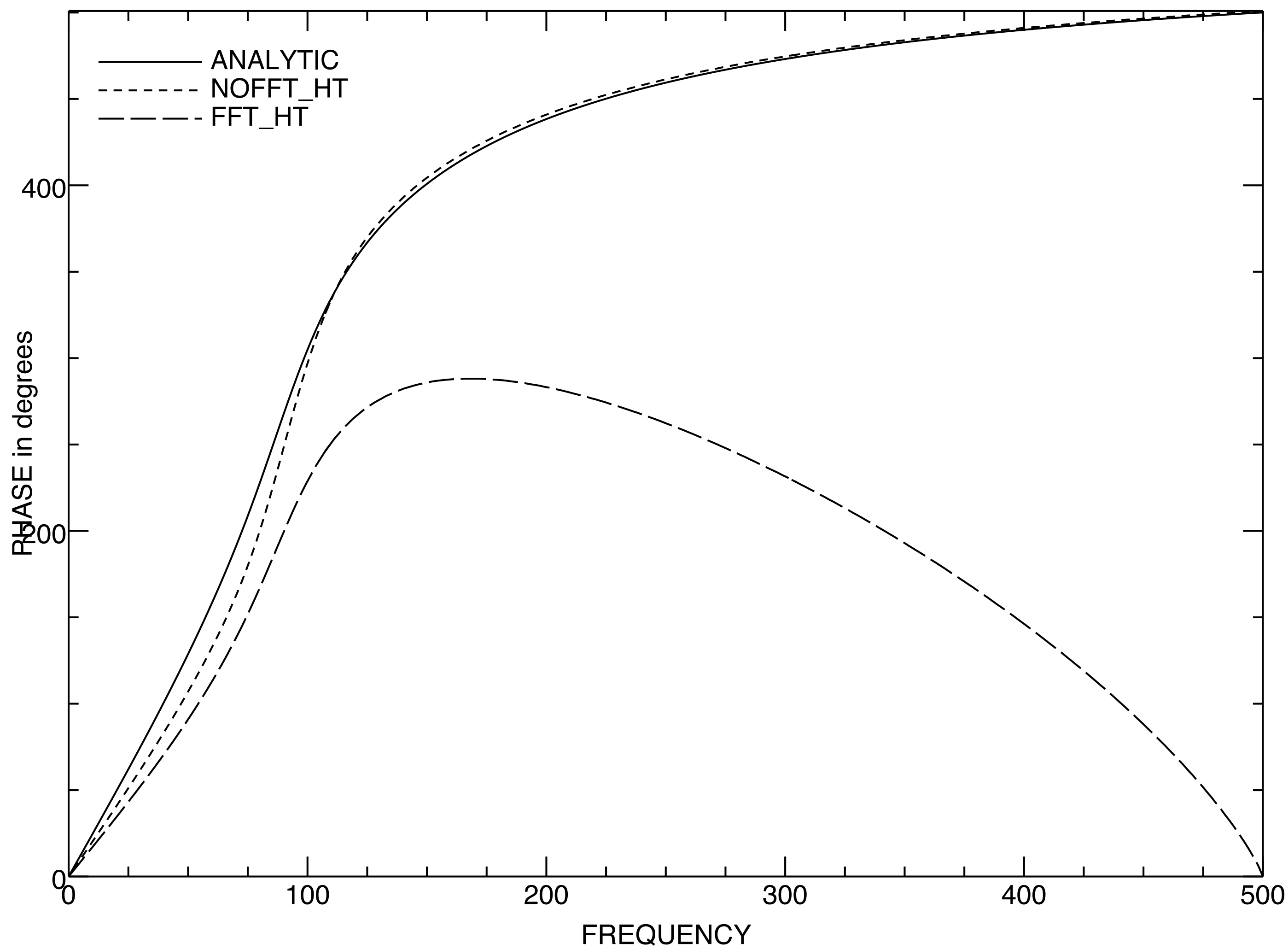
FIR filter falloff is VERY sharp



Dashed lines are 6-pole causal Butterworth low-pass filters.

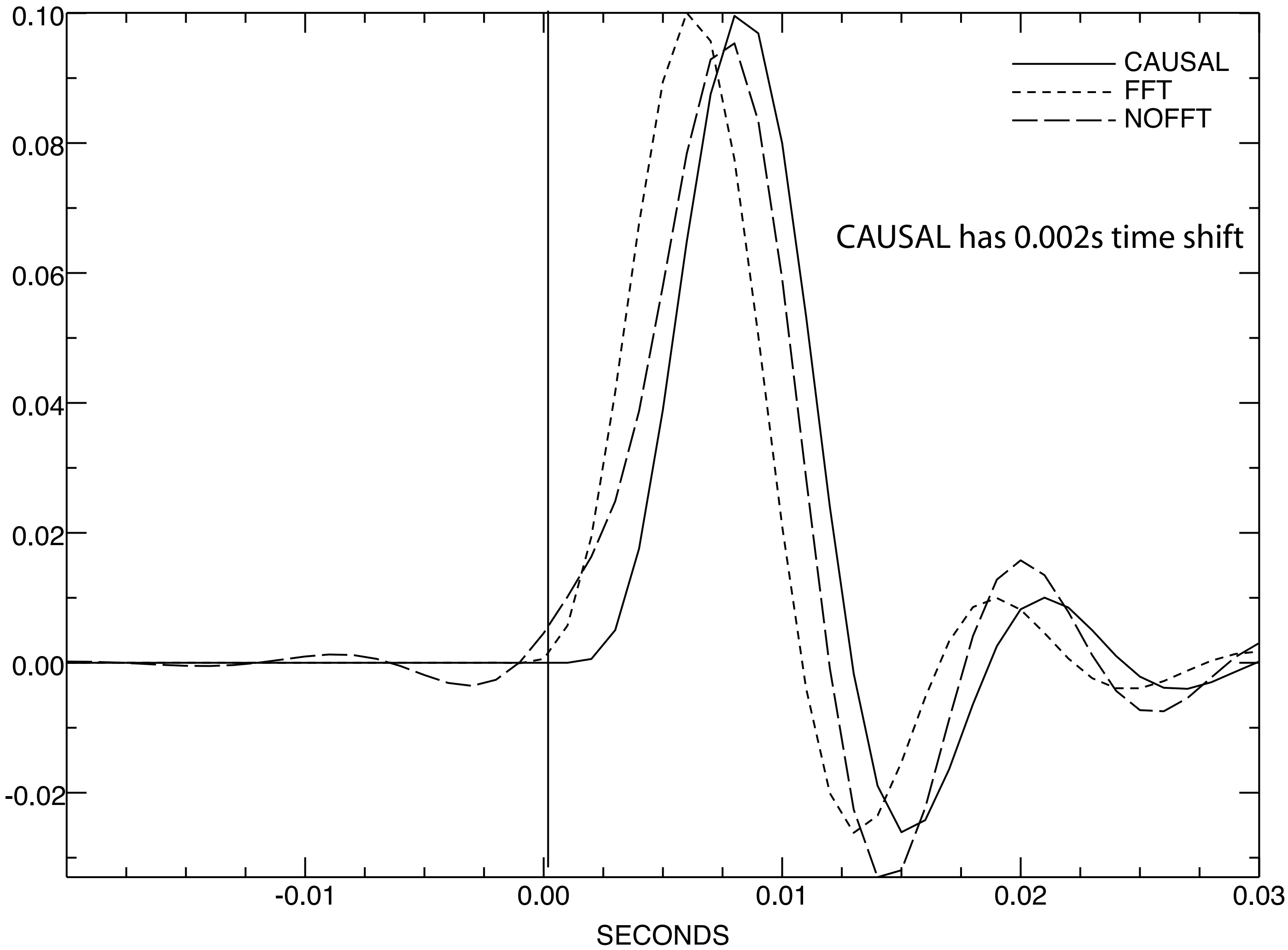
Choose 100 Hz corner because amplitude ≈FIR at Nyquist.

Digital FFT Phase $\rightarrow$ 0 at Nyquist



NOFFT agrees with ANALYTIC throughout the frequency range.

Hard to Choose Between FFT and NOFFT



NOFFT peak a better match for CAUSAL

FFT has no first-break time shift and peak closer to IMPULSE

FIR Filters for the Two Events

FIR filters forCalifornia microearthquake

- Input sample rate is 2.56×10^5 sps.
- Six stages of decimation/filtering
- Output sampling rate 1000 sps

FIR filters for VA aftershock

- Input sample rate is 1.024×10^5 sps.
- Eight stages of decimation/filtering
- Output sample rate 100 sps

Amplitudes calculated using program EVALRESP

FIR Filters for the Two Events

FIR filters forCalifornia microearthquake

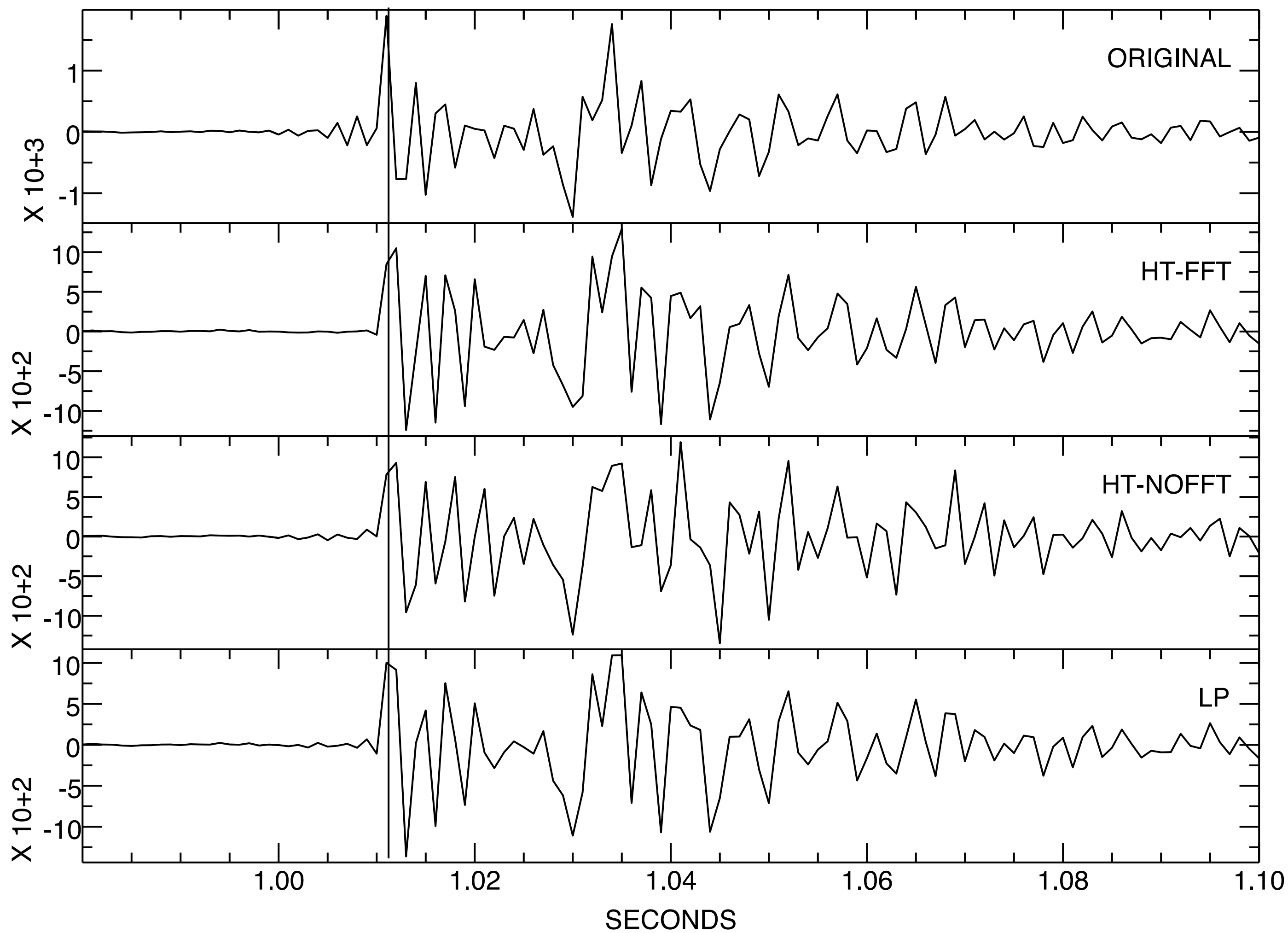
- Input sample rate is 2.56×10^5 sps.
- Six stages of decimation/filtering
  - Decimation factors: 8, 2, 2, 2, 2, 2
  - Coefficients: 33, 13, 13, 13, 13, 101
- Output sampling rate 1000 sps

FIR filters for VA aftershock

- Input sample rate is 1.024×10^5 sps.
- Eight stages of decimation/filtering
  - Decimation factors: 8, 2, 2, 2, 2, 2, 2, 2
  - Coefficients: 29, 13, 13, 13, 13, 13, 101, 95
- Output sample rate 100 sps

Amplitudes calculated using program EVALRESP

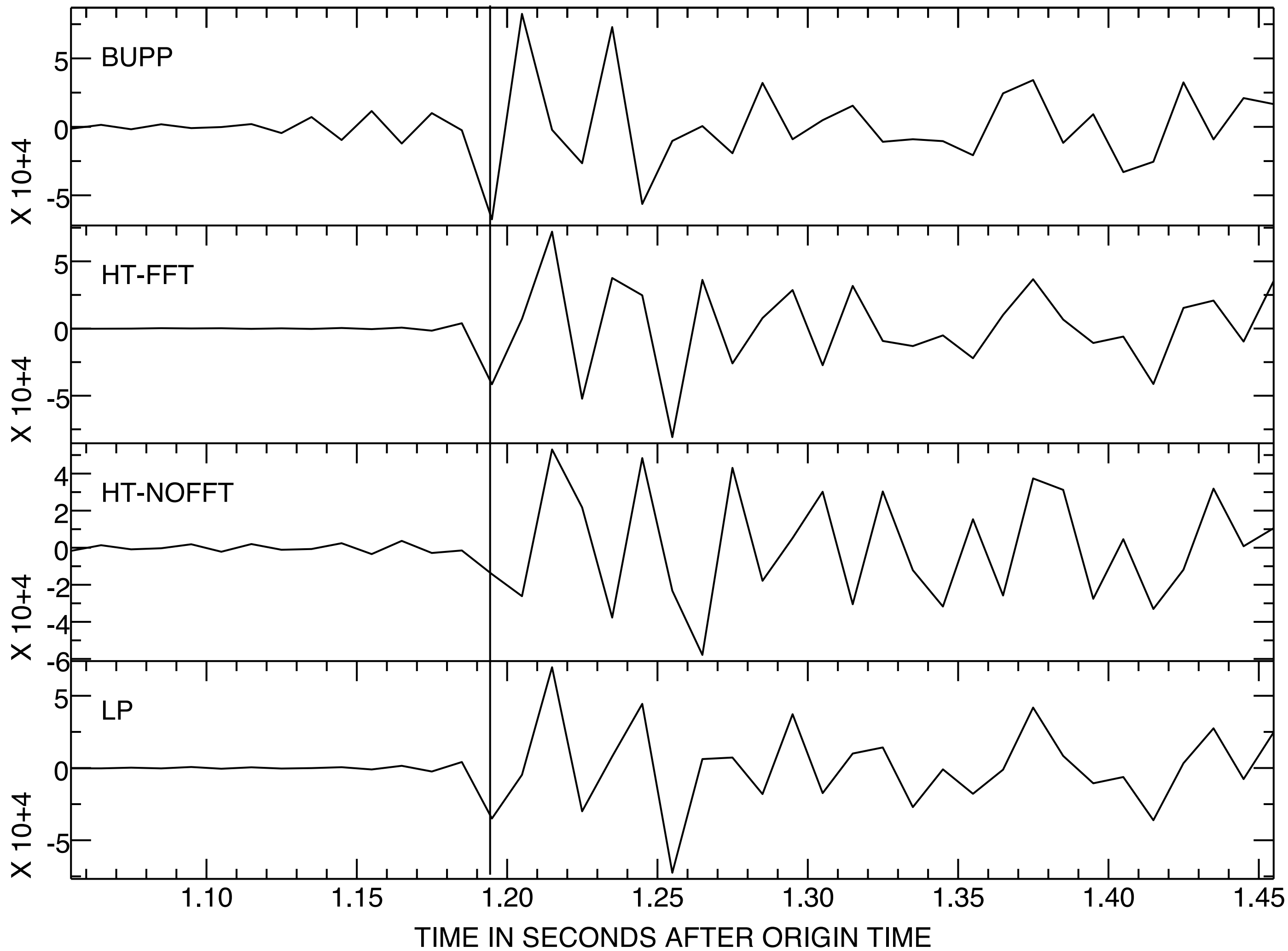
Microearthquake: Original and Processed



HT-NOFFT time shifted -0.002 s.

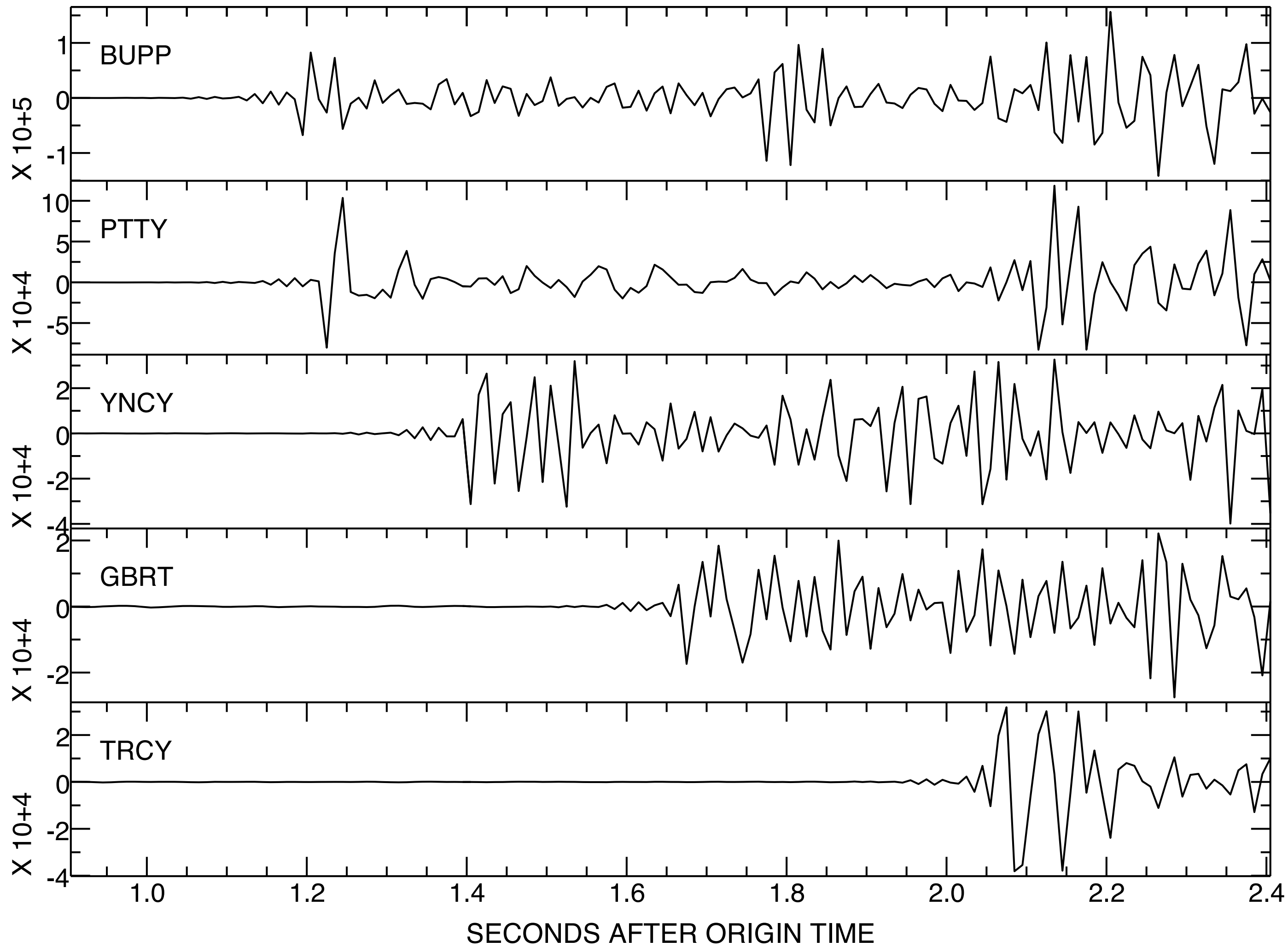
LP is SAC lp co 400 np 3 (Butterworth low-pass)

11/08/30 VA Aftershock: Original and Processed

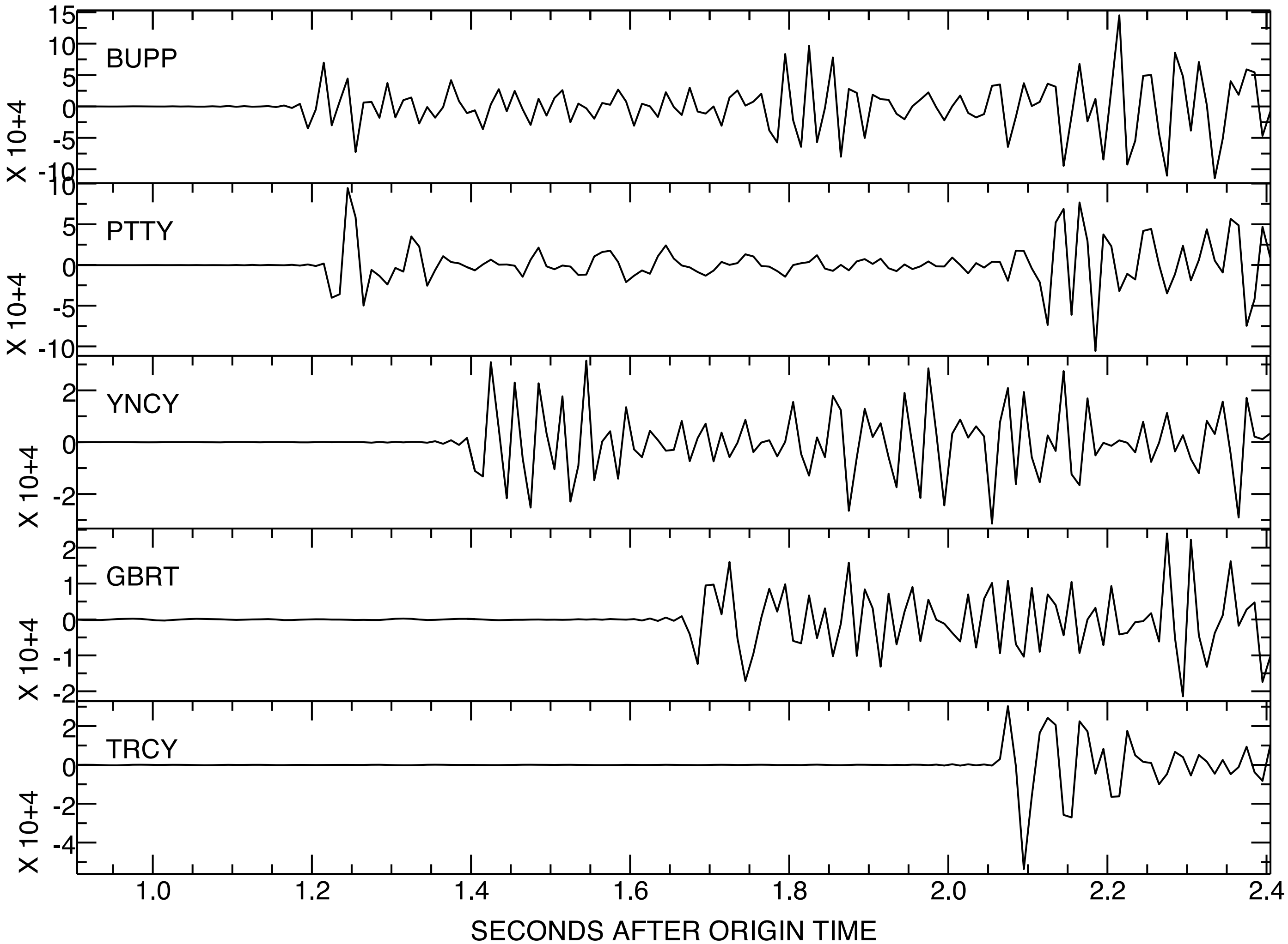


HT-NOFFT time shifted -0.02 s; LP corner 40 Hz

VA Earthquake Aftershock Raw



VA Earthquake Aftershock LP filtered



Conclusions

- Replacing a zero-phase FIR filter with a causal one calculated using the Hilbert Transform does "clean up" the waveform before an impulsive first arrival.
- FFT and NOFFT approaches do not give significantly different results
- The shape of the waveform is altered by either HT processing which may complicate detailed analysis (such as sub-events).
- Results from the HT techniques do not differ significantly from applying a Butterworth 3-pole causal low-pass filter with a corner at 80% the Nyquist frequency — a much simpler process than using a Hilbert Transform.