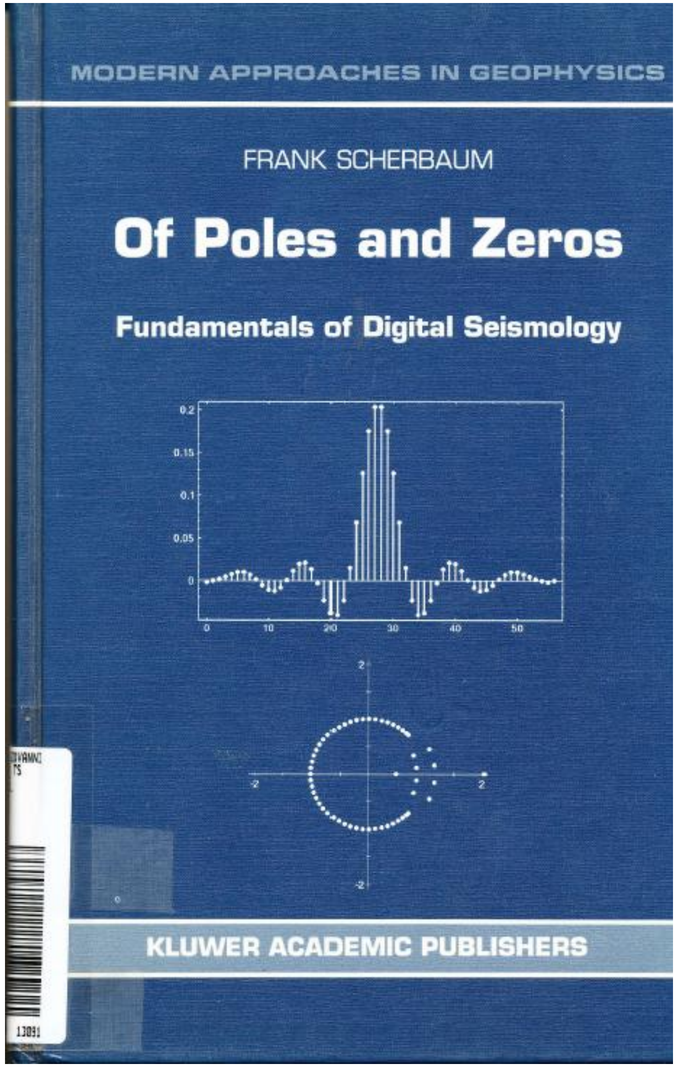
UNIVERSITÀ DEGLI STUDI DI TRIESTE

Applied seismology

Filters

Giovanni Costa - costa@units.it

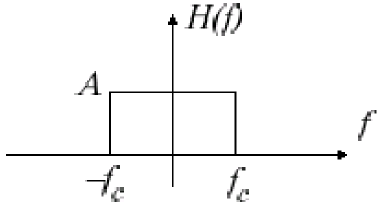

MODERN APPROACHES IN GEOPHYSICS
FRANK SCHERBAUM
Of Poles and Zeros
Fundamentals of Digital Seismology
KLUWER ACADEMIC PUBLISHERS

Filtri ideali

Sono di quattro tipi:

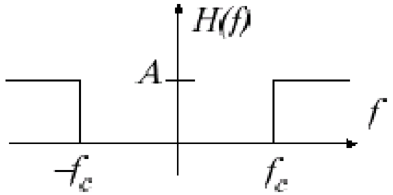
Passa basso

(f_c= *frequenza di taglio*)



Passa alto

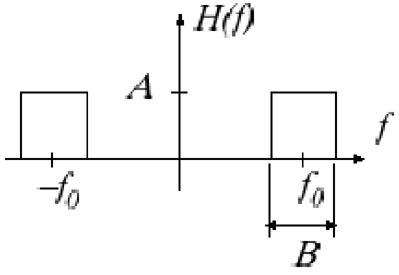
(f_c= *frequenza di taglio*)



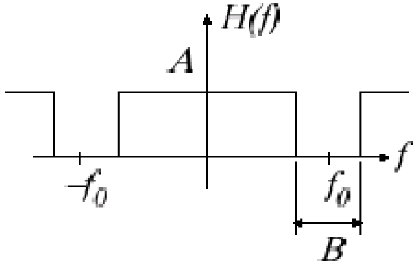
Passa banda

f_0 = *frequenza di riferimento (usualm. centrale)*

B= *banda passante*

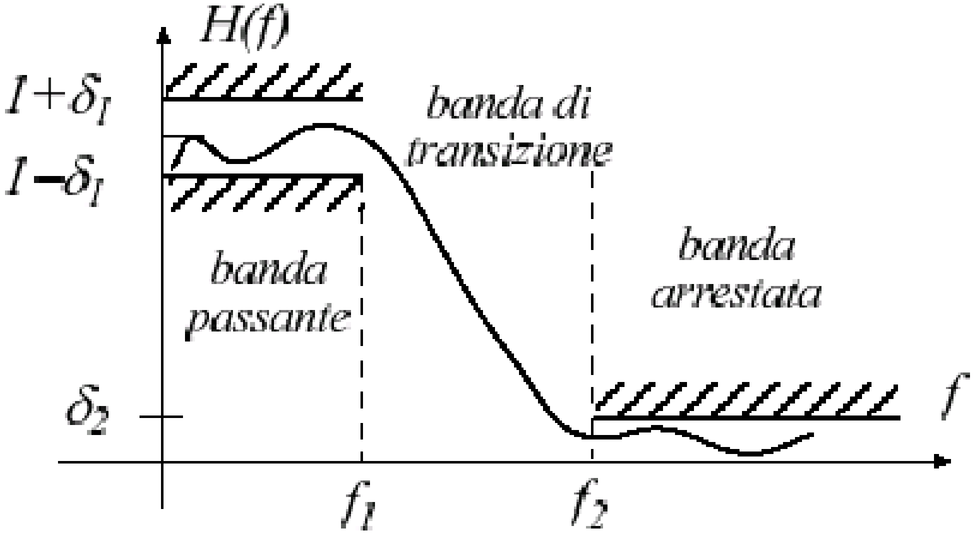


Arresta banda

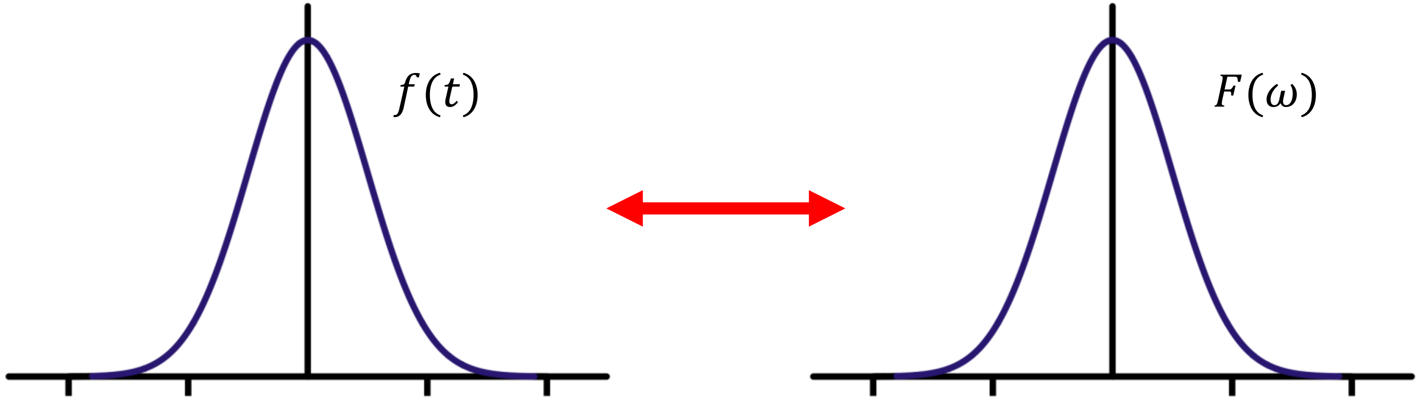


Real filters

Their frequency response is specified by masking:
Example (low pass):



Gaussian filter



$$H(\omega) = \exp\left(-\alpha\left(\frac{\omega-\omega_i}{\omega_i}\right)^2\right)$$

$$\alpha_{opt}(\omega_i) = \frac{\omega_i}{2}\left|\frac{d\tau_{gr}(\omega)}{d\omega}\right|_{\omega=\omega_i}$$

Filtro di Butterworth

$$\left|H_n(f)\right| = \frac{1}{\sqrt{1+\left(\frac{f}{f_c}\right)^{2n}}}$$

Oppure:

$$\left|H_n(\omega)\right|^2 = \frac{1}{1+\omega^{2n}}$$

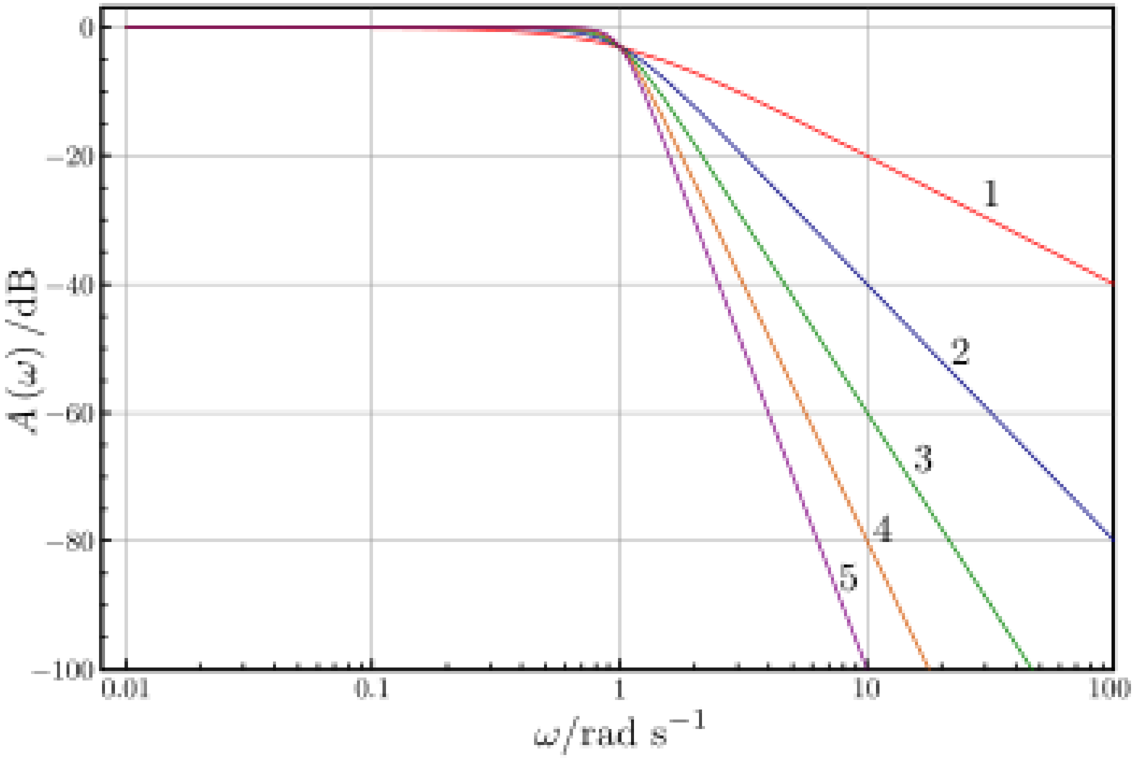
Con: $$\omega = \frac{2\pi f}{2\pi f_c}$$ Frequenza normalizzata

Butterworth Filters

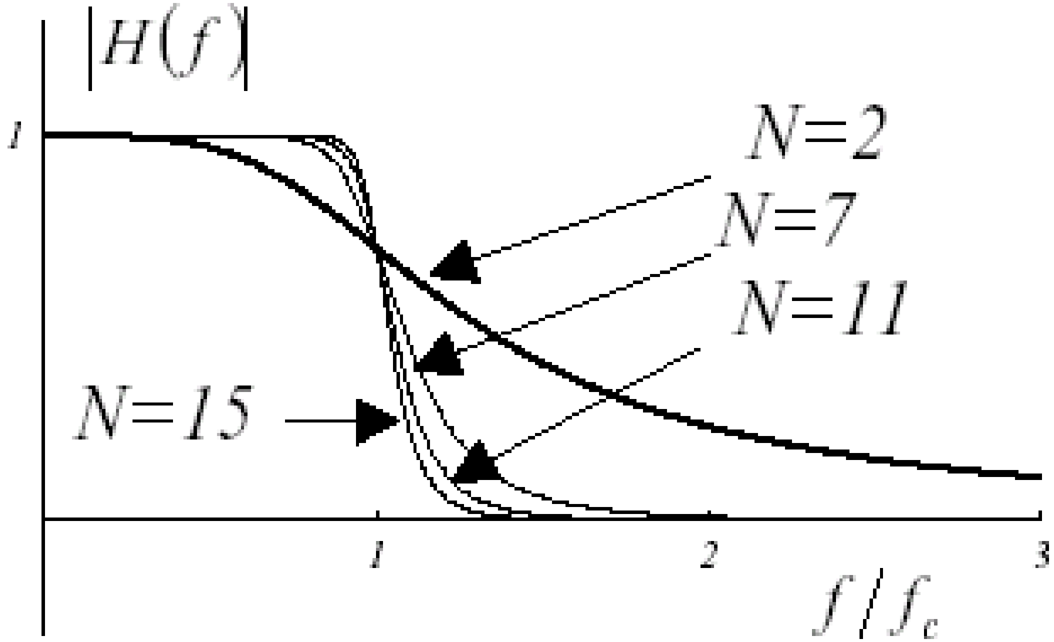
These are "maximally flat" filters in the passband, with frequency response.

$$\left|\frac{V_{out}}{V_{in}}\right| = \frac{1}{\sqrt{1+\left(\frac{f}{f_{3db}}\right)^{2n}}}$$

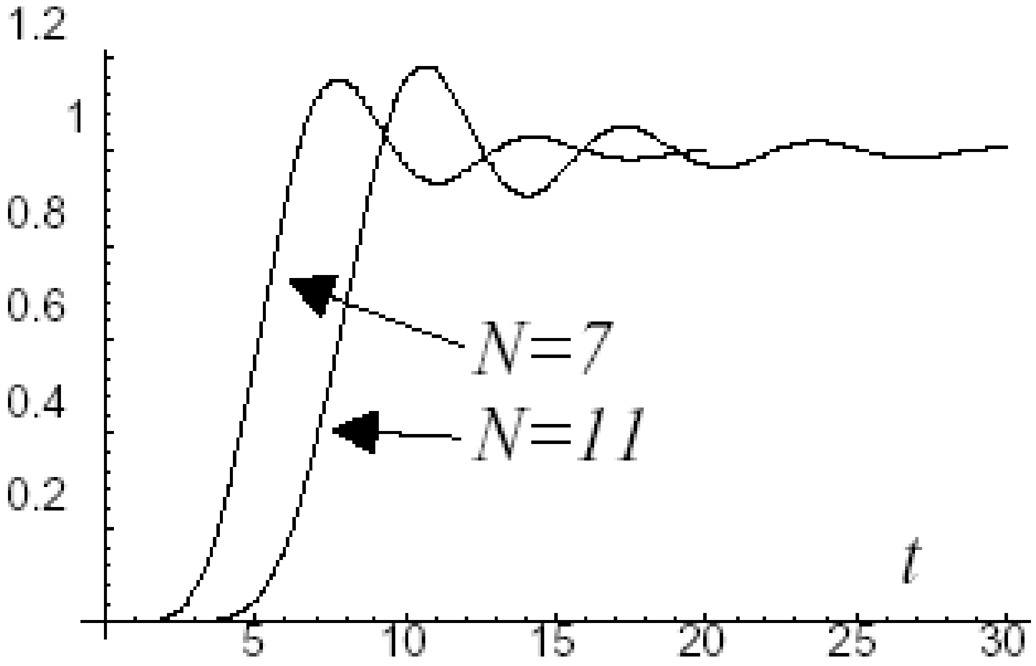
Where n is the order of the filter



Butterworth filters have the limitations of not being linear phase and requiring high order to ensure a sufficiently fast transition region.



step response



For modern seismic acquisition systems, the type of fllter required by the anti-aliasing filter is most easily implemented as an FIR (finite impulse response) filter. Some of the advantages and disadvantages of FIR and IIR (infinite response filter) are listed below.

FIR: Are always stable. Generally, several coefficients are required to obtainsteep filters. They can be causal and noncausal. One can easily construct linear phase or phase 0 filters.

IIR: They can potentially be unstable and subject to quantization errors. Steep filters are easily obtained with a small number of coefficients. Filtering therefore results in rapid results. Building 0-phase filters is difficult if not impossible. To remedy this, one can apply the filter twice in opposite directions.

Digital anti-aliasing commonly implemented in modern seismic acquisition systems belong to the generalized class of linear phase filters that do not cause phase distortion of the signal. In general, they produce a constant time shift that can be corrected. If the time shift is zero or is corrected, the filter would be called a zero-phase filter. The reasons for using FIR filters in this context are mainly based on the fact that there are established design procedures to fit the required specifications with great precision, requiring very steep ramps, (these would be potentially unstable as IIR filters), the linear phase property can only be implemented exactly as FIR fllters. IIR filters always cause phase distortion within the filter passband

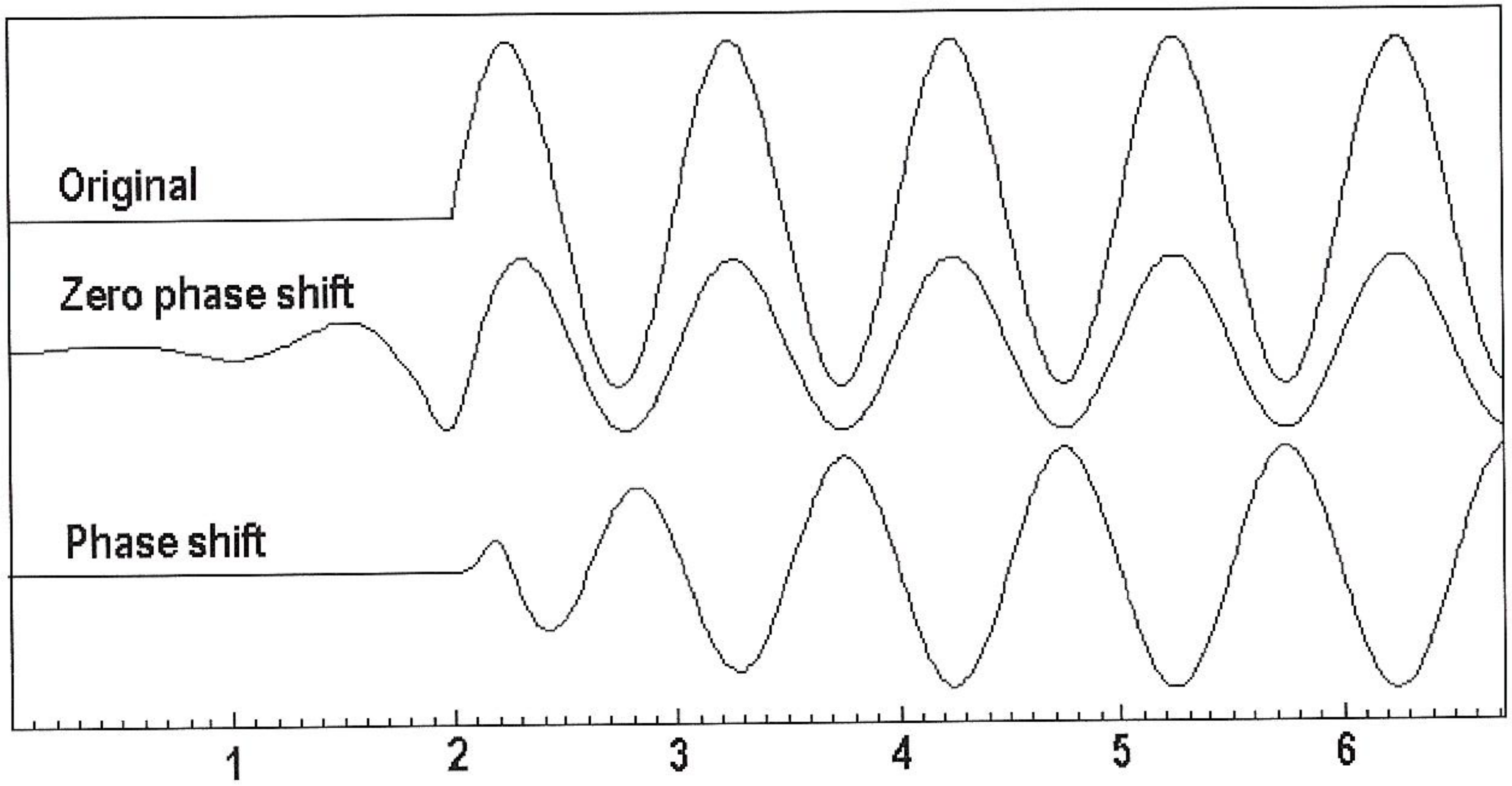
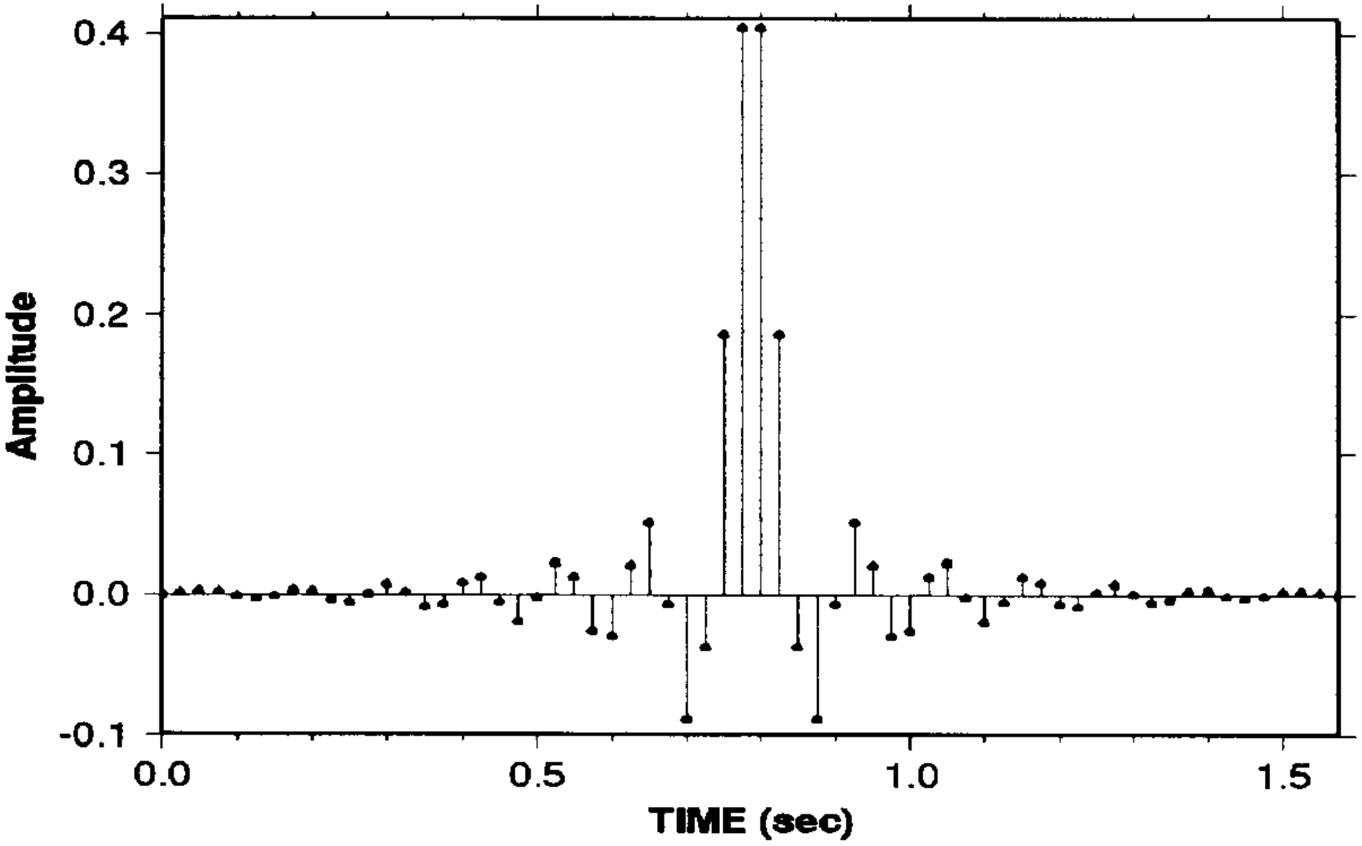


Figure 6.12 The effect of using filters on a signal with a sharp onset. The signal is a 1 Hz sin wave with an onset at 2 s (top trace). The middle trace shows the signal filtered with a zero phas shift Butterworth filter while the bottom trace shows the signal filtered with a Butterworth filter with phase shift. Both filters are band pass with corners at 1 and 5 Hz.

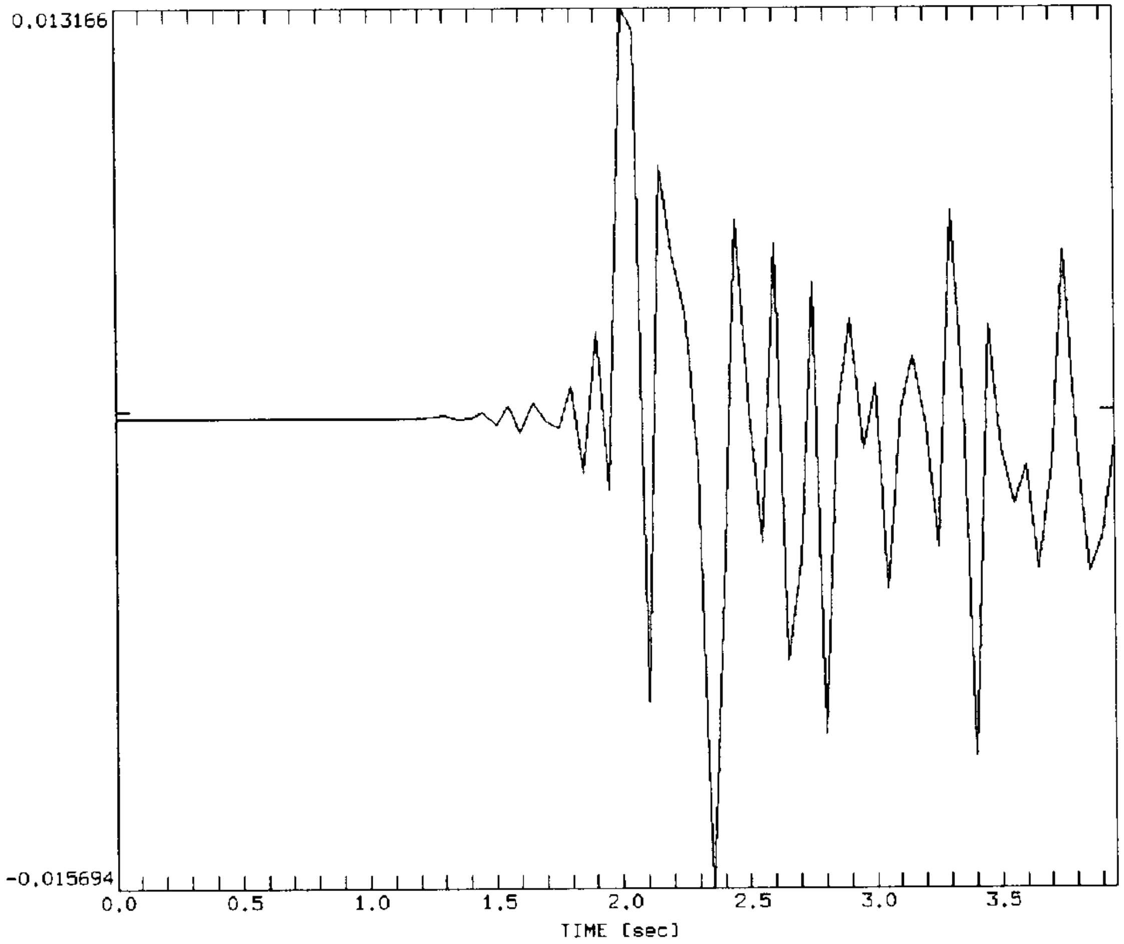
Filtro **FIR** anti alias
Quanterra 40 Hz data stream - decimazione a 20 Hz



Filtro impulse response FIR dello stage 4 del digitalizzatore QDP 380 di Quanterra. E’ usato come filtro lineare digitale anti-alias nellstream a 40 Hz per la decimazione a 20 Hz.

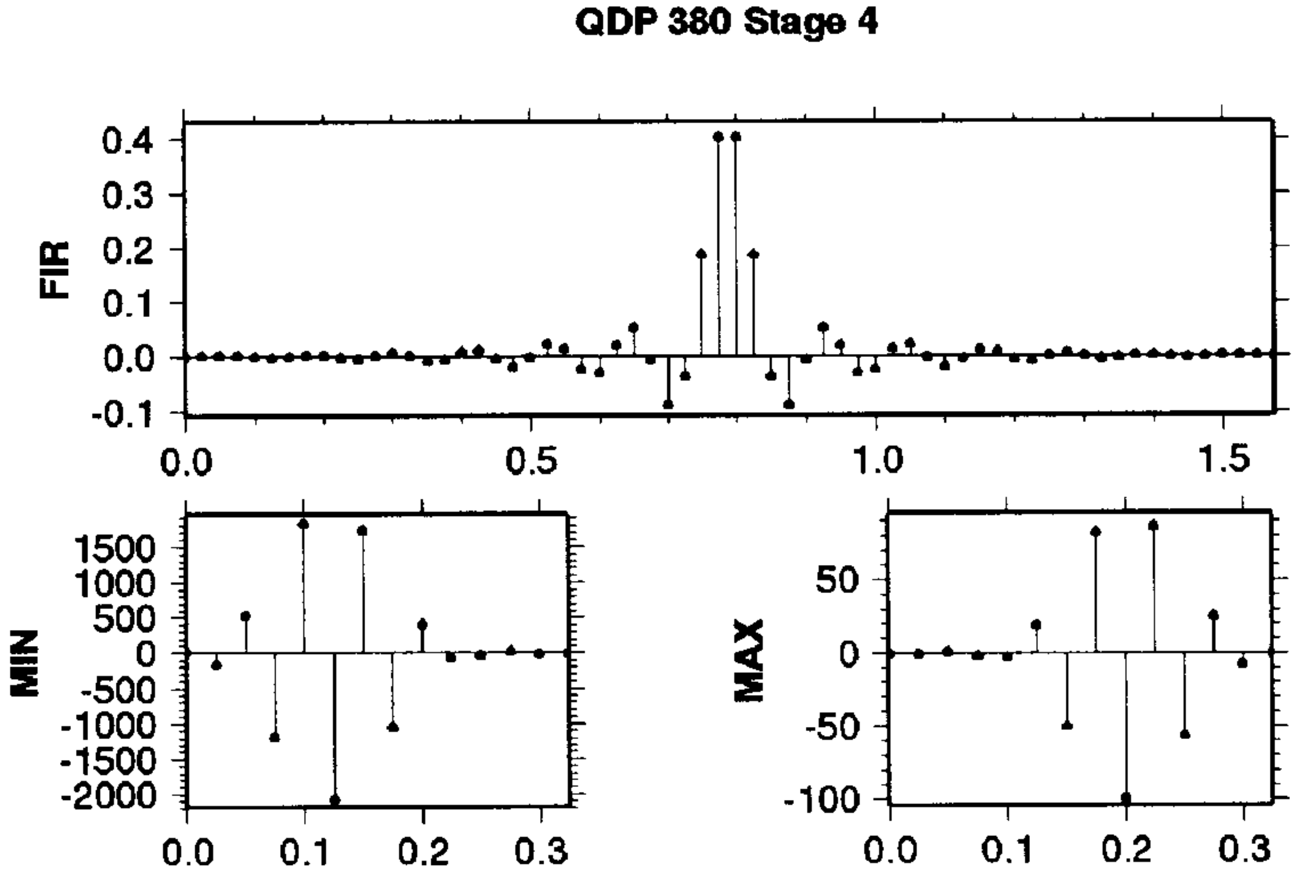
Such filters with symmetrical impulse responses are often called bifacial or acausal. The impulse response oscillates with frequencies close to the corner frequency of the filter, an effect called the Gibb phenomenon. It can be reduced at the cost of filter passband width by decreasing the slope of the flltro. This, however, would also reduce the range the frequency usable by the signal.

As a consequence of the symmetry of the filter impulse response, the onset of the very impulsive signals can be obscured by oscillations, precursors ("acausal"), and become difficult, if not impossible, to determine.



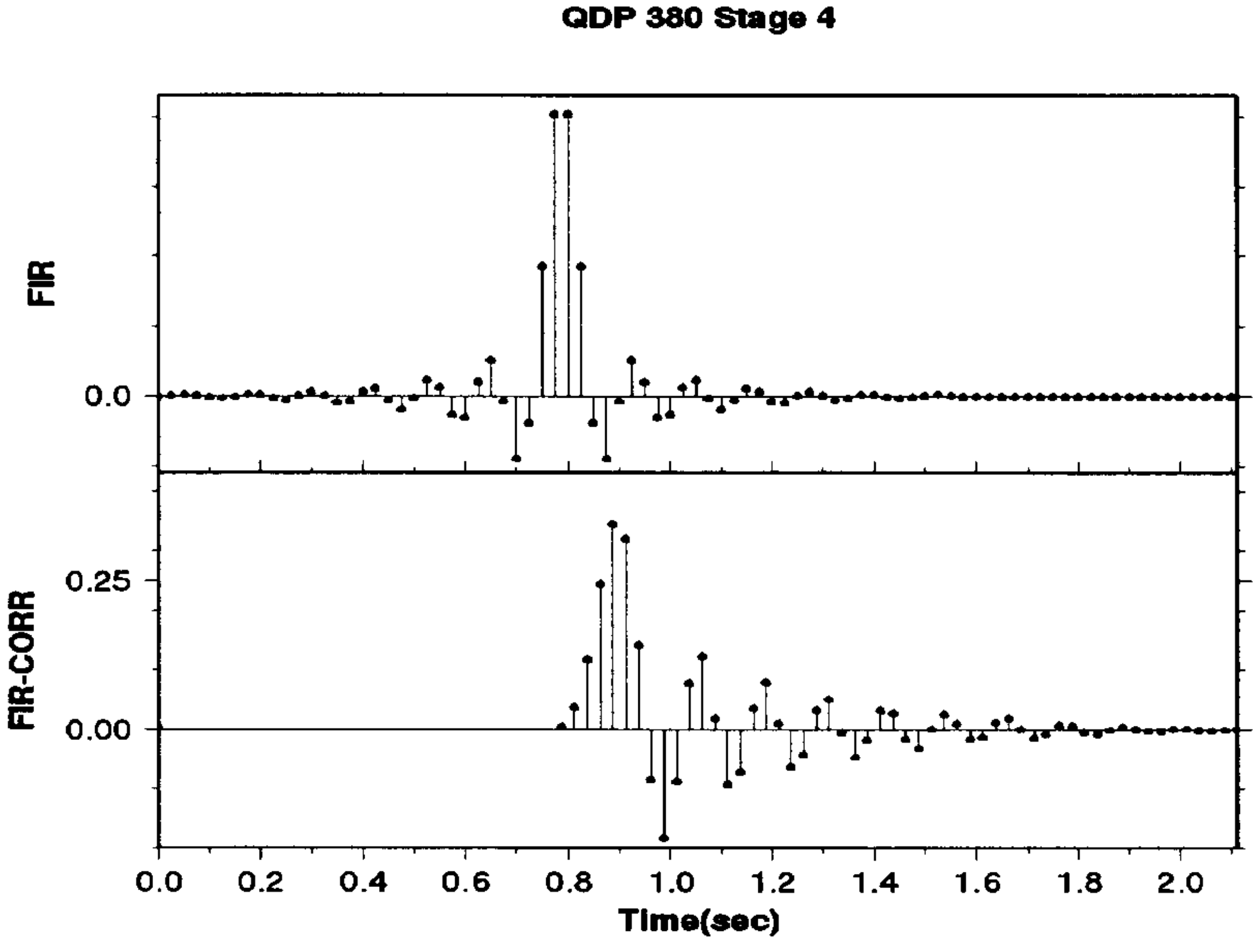
The trace shows the portion of the P wave on the vertical component sampled at 20 Hz. Before decimation from 40 to 20 Hz it was filtered with the FIR filter. As can be seen, the use of a symmetric FIR filter in this case considerably obscures the seismic onset and generates spurious acusality.

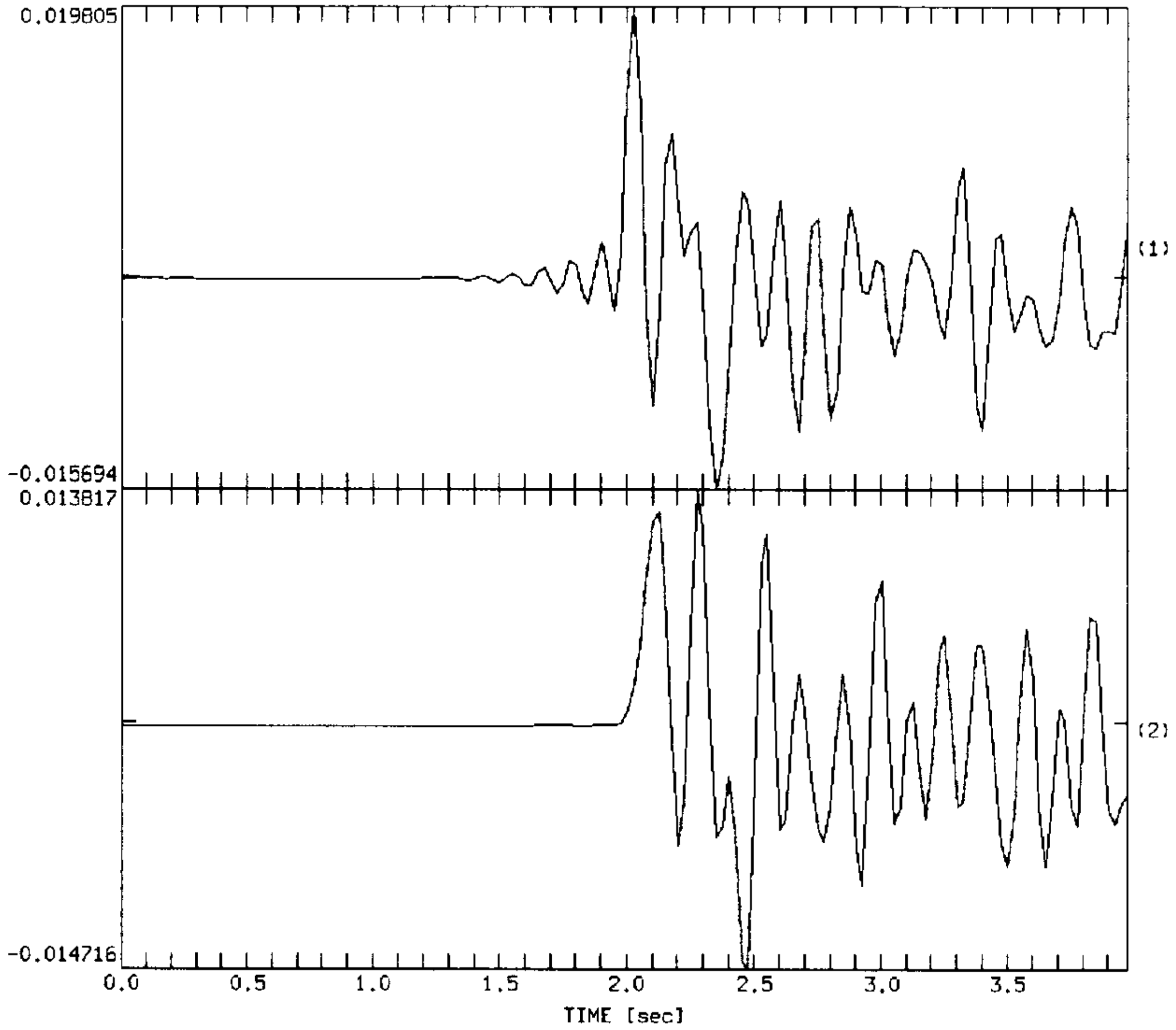
One way to deal with this problem, which is discussed below, is to construct a filter that removes the acausal component (left side) of the filtered signal and replaces it with its equivalent causal component (right side).



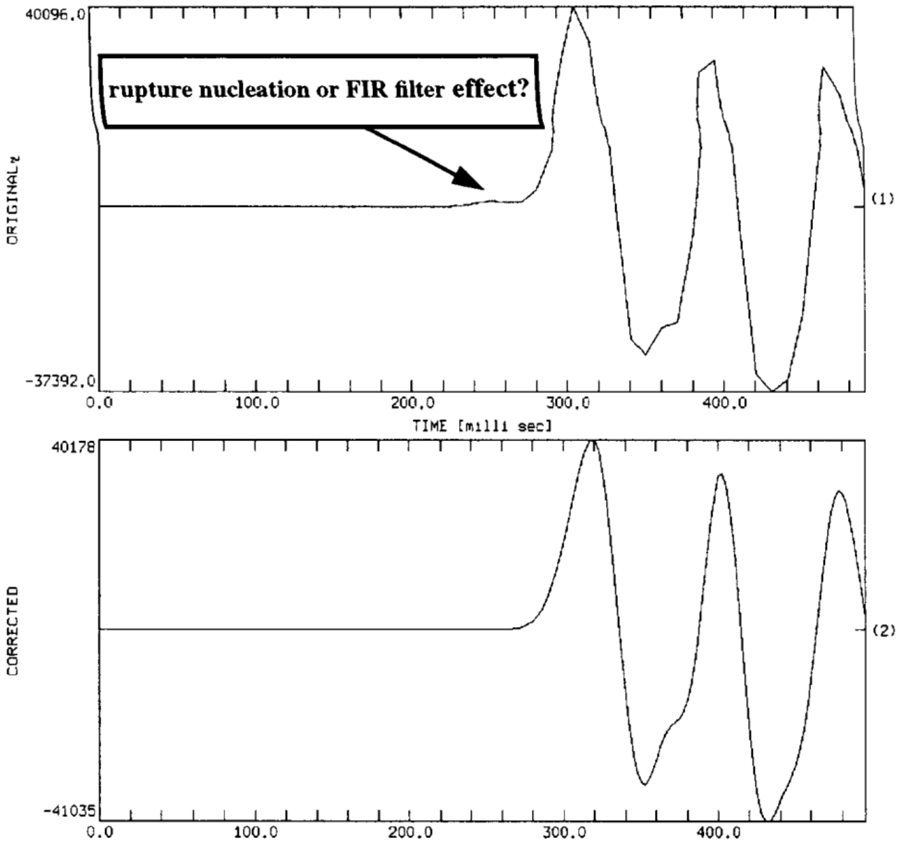
The impulse response of the linearFIR phase filter can be separated into its minimum and maximum components. The left or 'acausal' part is caused by the maximum phase component, while the 'causal' part on the right side responds to the minimum phase component. The full transfer function of the filter is the product of the two. The solution is to remove the maximum phase component from the filter output trace without changing the amplitude characteristic value of the filter process.

Replacing the maximum phase component of the FIR filter response with its equivalent minimum phase. The top trace shows the original FIR filter response (with zeros added). The bottom trace shows the result of the correction. In addition, a time shift was applied to the bottom trace to correct the linear phase component of the FIR filter response.

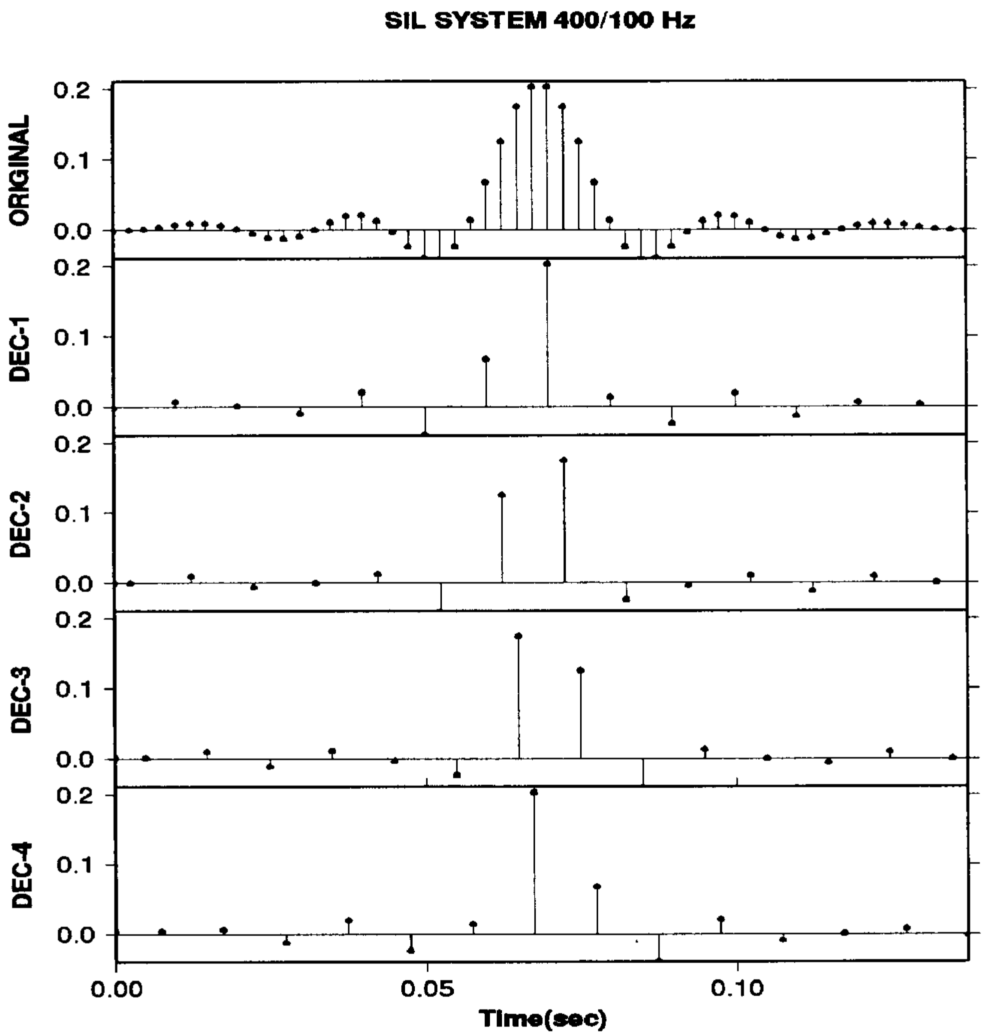




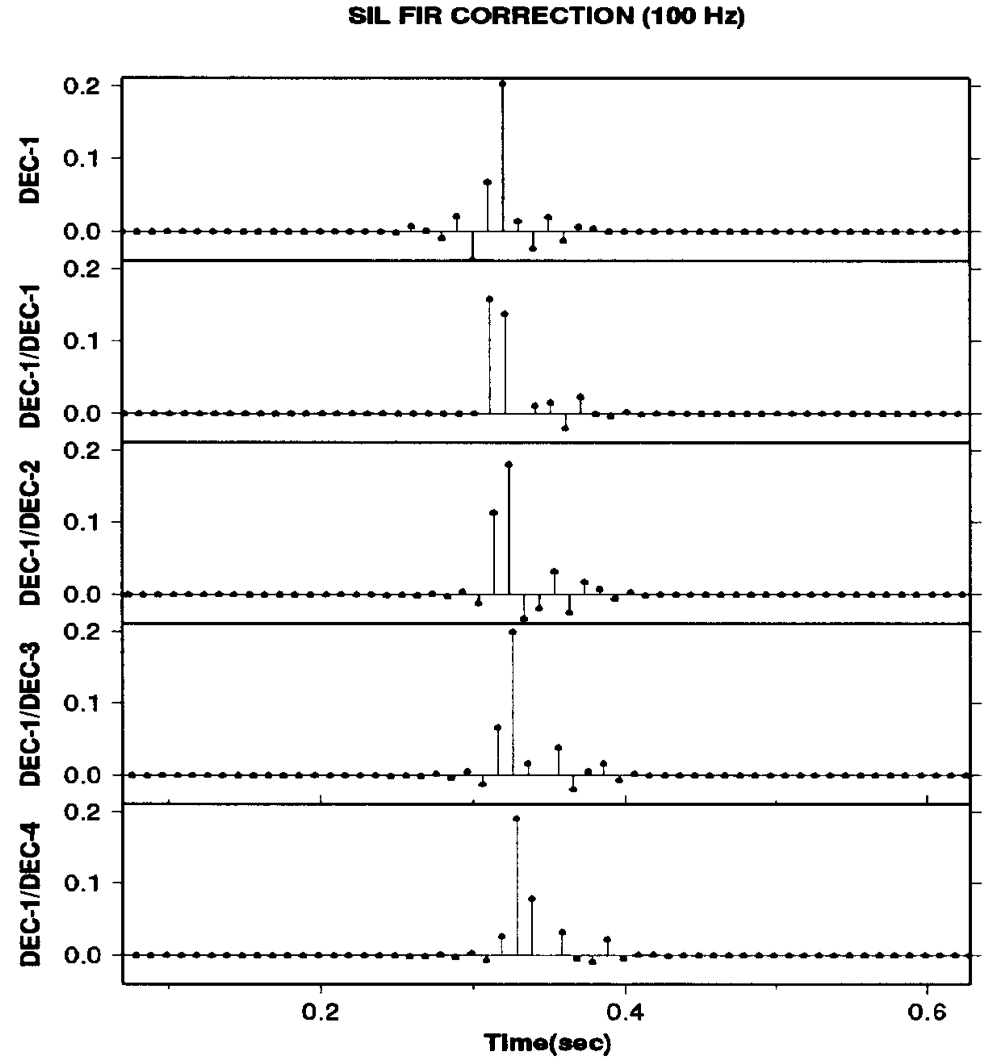
Removal of the maximum phase component of the FIR filter response from the signal. The top trace shows the interpolated and resampled signal at 40 Hz. The bottom trace shows the corrected signal.



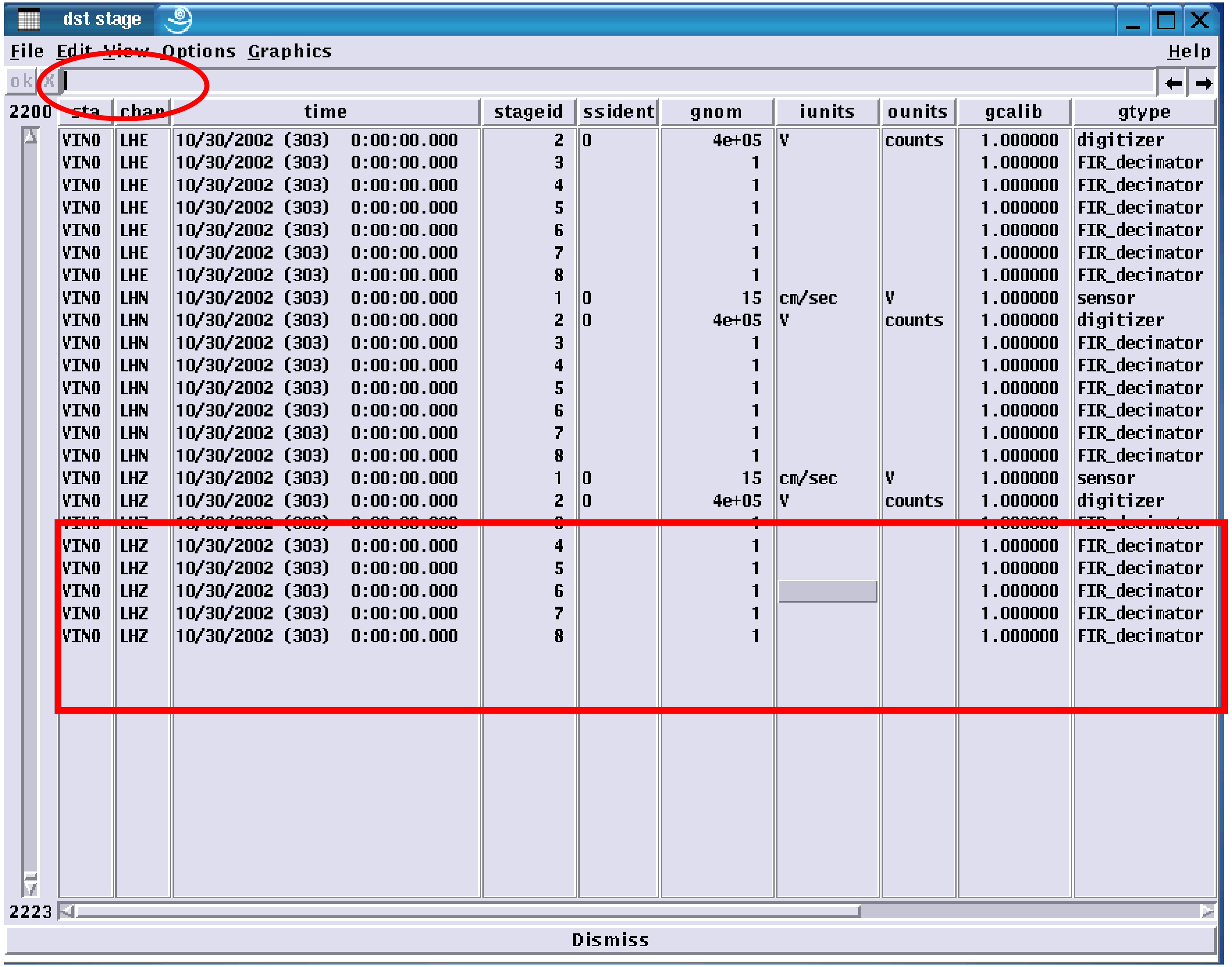
Vertical component of P-wave arrival for a local event recorded at the HEI station of the SIL network in southern Iceland. Decimated data sampled at 100 Hz in the upper trace are shown. Note the precursor signal. The bottom trace shows the corrected trace after interpolation at 400 Hz and removal of the maximum phase component of the FIR filter response of the SIL system.



FIR filter impulse response for the SIL system for the decimation step 400 at 100 Hz. The original response sampled at 400 Hz is shown in the upper trace. The different possible decimation responses sampled at 100 Hz are shown below. The DEC-1 decimation scheme includes samples 1,5,9,13, ".; DEC-2 samples 2,6,10,14, ...; DEC-3 samples 3,7,11,15, ... and DEC-4 samples 4,8'12'16 '... Of the original answer.



Removal of the maximum phase component from the decimated filtered trace. The upper trace shows the decimated FIR filter response. This was used as the signal. Traces 2 through 5 show the result of applying the calculated correction filters for the four DEC1-DEC4 decimation schemes.



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VINO	LHE	10/30/2002 (303) 0:00:00.000	3		1			1.000000	FIR_decimator
VINO	LHE	10/30/2002 (303) 0:00:00.000	4		1			1.000000	FIR_decimator
VINO	LHE	10/30/2002 (303) 0:00:00.000	5		1			1.000000	FIR_decimator
VINO	LHE	10/30/2002 (303) 0:00:00.000	6		1			1.000000	FIR_decimator
VINO	LHE	10/30/2002 (303) 0:00:00.000	7		1			1.000000	FIR_decimator
VINO	LHE	10/30/2002 (303) 0:00:00.000	8		1			1.000000	FIR_decimator
VINO	LHN	10/30/2002 (303) 0:00:00.000	1	0	15	cm/sec	V	1.000000	sensor
VINO	LHN	10/30/2002 (303) 0:00:00.000	2	0	4e+05	V	counts	1.000000	digitizer
VINO	LHN	10/30/2002 (303) 0:00:00.000	3		1			1.000000	FIR_decimator
VINO	LHN	10/30/2002 (303) 0:00:00.000	4		1			1.000000	FIR_decimator
VINO	LHN	10/30/2002 (303) 0:00:00.000	5		1			1.000000	FIR_decimator
VINO	LHN	10/30/2002 (303) 0:00:00.000	6		1			1.000000	FIR_decimator
VINO	LHN	10/30/2002 (303) 0:00:00.000	7		1			1.000000	FIR_decimator
VINO	LHN	10/30/2002 (303) 0:00:00.000	8		1			1.000000	FIR_decimator
VINO	LHZ	10/30/2002 (303) 0:00:00.000	1	0	15	cm/sec	V	1.000000	sensor
VINO	LHZ	10/30/2002 (303) 0:00:00.000	2	0	4e+05	V	counts	1.000000	digitizer
VINO	LHZ	10/30/2002 (303) 0:00:00.000	3		1			1.000000	FIR_decimator
VINO	LHZ	10/30/2002 (303) 0:00:00.000	4		1			1.000000	FIR_decimator
VINO	LHZ	10/30/2002 (303) 0:00:00.000	5		1			1.000000	FIR_decimator
VINO	LHZ	10/30/2002 (303) 0:00:00.000	6		1			1.000000	FIR_decimator
VINO	LHZ	10/30/2002 (303) 0:00:00.000	7		1			1.000000	FIR_decimator
VINO	LHZ	10/30/2002 (303) 0:00:00.000	8		1			1.000000	FIR_decimator

dst stage

ile Edit View Options Graphics Help

:200 stageid	ssident	gnom	iunits	ounits	gcalib	gtype	decifac	samprate	dir	dfile
2	0	4e+05	V	counts	1.000000	digitizer		2000.000000		
3		1			1.000000	FIR_decimator	5	400.0000000	response/stage	FS2D5M
4		1			1.000000	FIR_decimator	2	200.0000000	response/stage	F96CM
5		1			1.000000	FIR_decimator	2	100.0000000	response/stage	F96CM
6		1			1.000000	FIR_decimator	5	20.0000000	response/stage	FS2D5
7		1			1.000000	FIR_decimator	2	10.0000000	response/stage	F96C
8		1			1.000000	FIR_decimator	10	1.0000000	response/stage	F260
1	0	15	cm/sec	V	1.000000	sensor			response/stage	cmg3t_pz
2	0	4e+05	V	counts	1.000000	digitizer		2000.000000		
3		1			1.000000	FIR_decimator	5	400.0000000	response/stage	FS2D5M
4		1			1.000000	FIR_decimator	2	200.0000000	response/stage	F96CM
5		1			1.000000	FIR_decimator	2	100.0000000	response/stage	F96CM
6		1			1.000000	FIR_decimator	5	20.0000000	response/stage	FS2D5
7		1			1.000000	FIR_decimator	2	10.0000000	response/stage	F96C
8		1			1.000000	FIR_decimator	10	1.0000000	response/stage	F260
1	0	15	cm/sec	V	1.000000	sensor			response/stage	cmg3t_pz
2	0	4e+05	V	counts	1.000000	digitizer		2000.000000		
3		1			1.000000	FIR_decimator	5	400.0000000	response/stage	FS2D5M
4		1			1.000000	FIR_decimator	2	200.0000000	response/stage	F96CM
5		1			1.000000	FIR_decimator	2	100.0000000	response/stage	F96CM
6		1			1.000000	FIR_decimator	5	20.0000000	response/stage	FS2D5
7		1			1.000000	FIR_decimator	2	10.0000000	response/stage	F96C
8		1			1.000000	FIR_decimator	10	1.0000000	response/stage	F260

:223

Dismiss

```
# FS2D5M Quanterra FIR filter parameter file
# 5 decimation factor
# delay=11.641
# type                  2
# npoles          0
# input sample interval 1.0e-03
# decimal factor  5
# normalization factor  1.0
# gain factor           1.0
theoretical  1  anti_alias   fir    coef_conv
1000 5
160
 1.335380875E-05  0.0
-3.923332770E-06  0.0
-7.131475286E-06  0.0
-1.059135775E-05  0.0
-1.172398970E-05  0.0
-8.714509931E-06  0.0
-5.306039839E-07  0.0
 1.049249931E-05  0.0
 2.181281889E-05  0.0
 2.850974306E-05  0.0
 2.577766281E-05  0.0
 1.271369365E-05  0.0
-9.378560208E-06  0.0
-3.531222319E-05  0.0
-5.528407564E-05  0.0
```

```
# UCSD Response file
#
# @(#)sts2_vel     1.4 3/5/94
# STS-2     8.00000E-03 hz seimometer
# Normalized  response relative to velocity
# 1                    type
# 2                    num of zeroes
# 2                    num of poles
# 0.0                  input sample interval
# 1                    decim factor
# 5.92E+07             normalization factor
# 1.0                  gain
theoretical    1 anti-alias    paz    pz6seismo
5.9200000000e+07 0.0000000000e+00
       5
-3.5540000000e-02 3.5540000000e-02 0.0000000000e+00 0.0000000000e+00
-3.5540000000e-02 -3.5540000000e-02 0.0000000000e+00 0.0000000000e+00
-2.5130000000e+02 0.0000000000e+00 0.0000000000e+00 0.0000000000e+00
-1.3100000000e+02 4.6730000000e+02 0.0000000000e+00 0.0000000000e+00
-1.3100000000e+02 -4.6730000000e+02 0.0000000000e+00 0.0000000000e+00
       2
0.0000000000e+00 0.0000000000e+00 0.0000000000e+00 0.0000000000e+00
0.0000000000e+00 0.0000000000e+00 0.0000000000e+00 0.0000000000e+00
# Qx80 Quanterra FIR filter parameter file
# Nikolaus Horn, 23.5.2001
# using information from Bob Busby
# stages used in 1992/08/12 firmware
# stage 1 of 5120/320/80/40/20/10/1
#
# input sample interval 5120
# decimation factor 16
# number of coefficients      64
# normalization factor  1.0
# gain factor           1.0
theoretical    2 anti_alias    fir    qdpk805
5.1200000000e+03    16
      64
-1.1132811200e-03 0.0000000000e+00
-1.0080020900e-03 0.0000000000e+00
-1.3528608200e-03 0.0000000000e+00
-1.7304536900e-03 0.0000000000e+00
-2.0841800100e-03 0.0000000000e+00
-2.3853771800e-03 0.0000000000e+00
-2.6095563000e-03 0.0000000000e+00
-2.7335225600e-03 0.0000000000e+00
-2.7331619000e-03 0.0000000000e+00
-2.5847244500e-03 0.0000000000e+00
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```
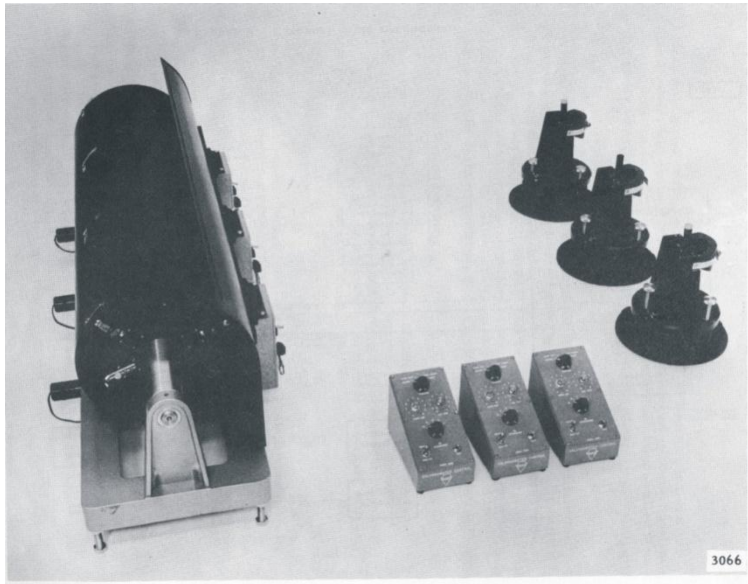
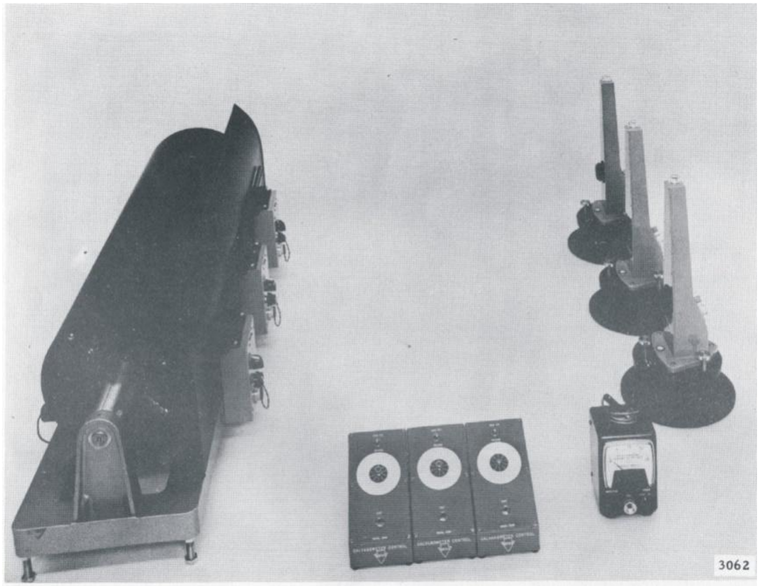

Wood-Anderson 1922





Short period (on the left) and long period (on the right) seismograph recording instruments

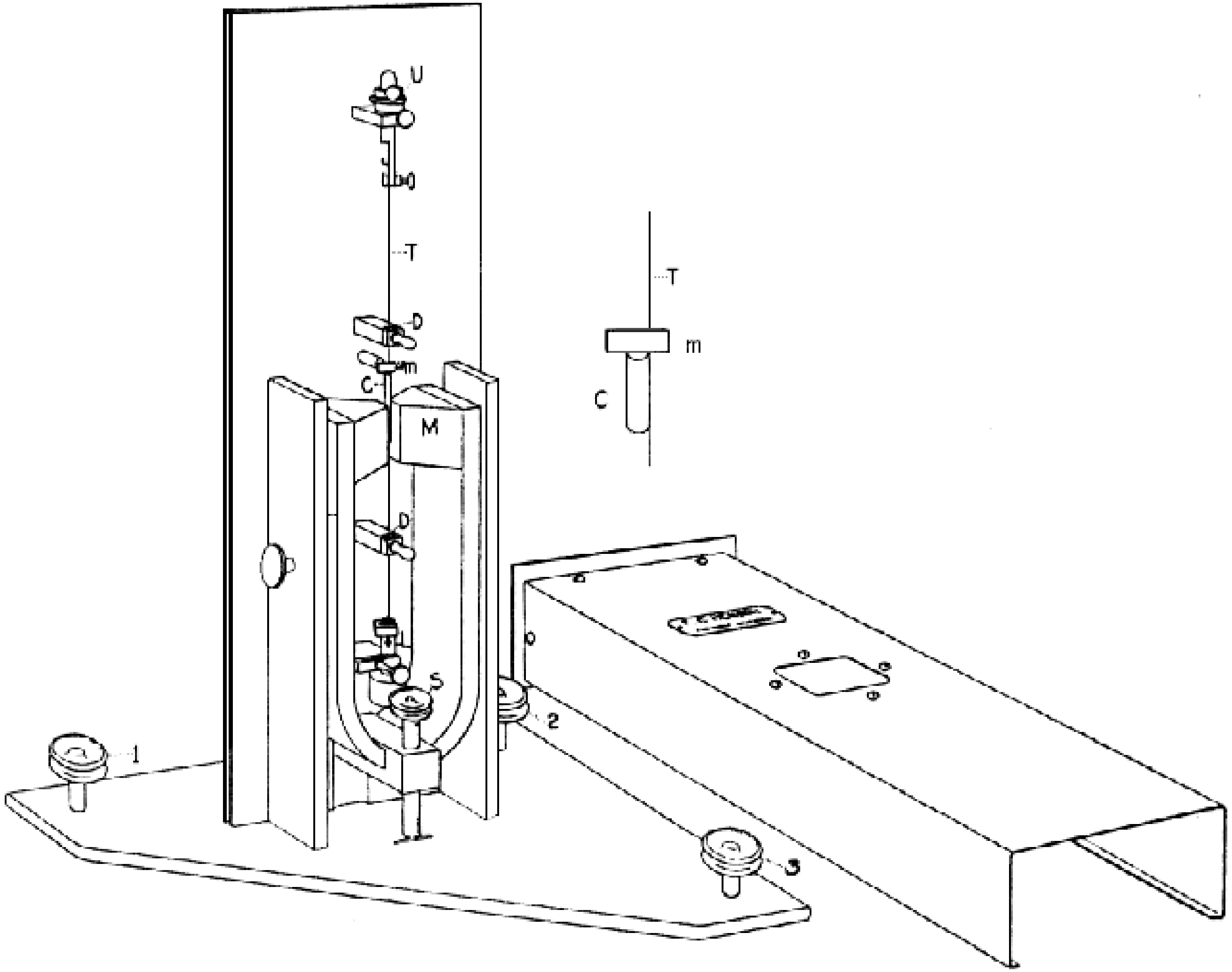
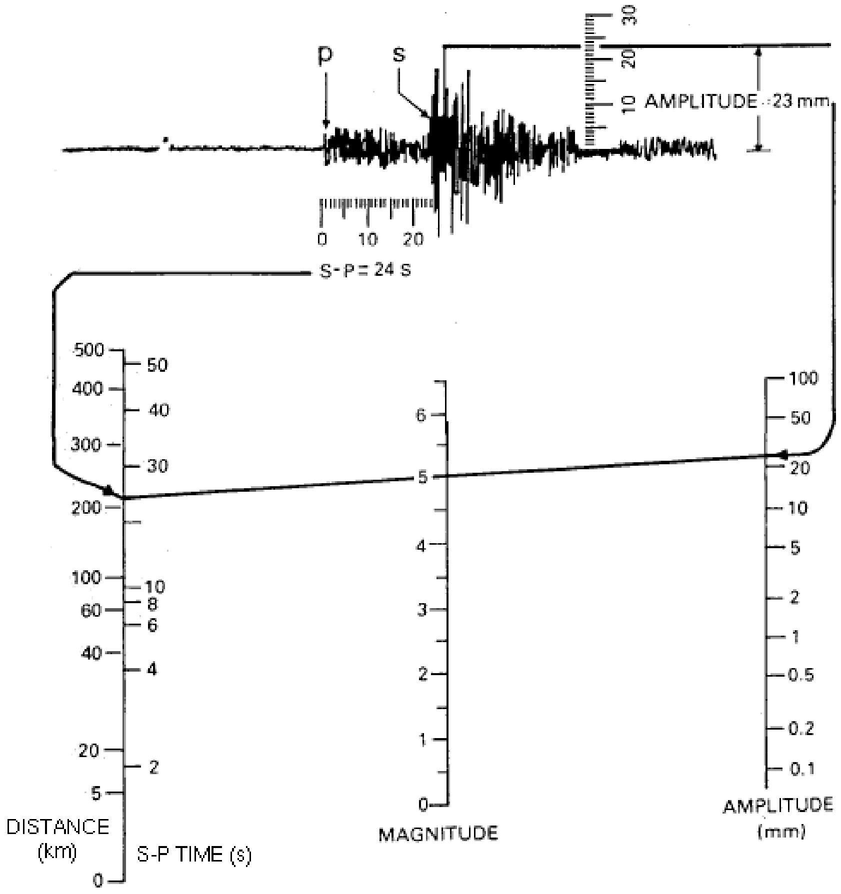


Fig. 2

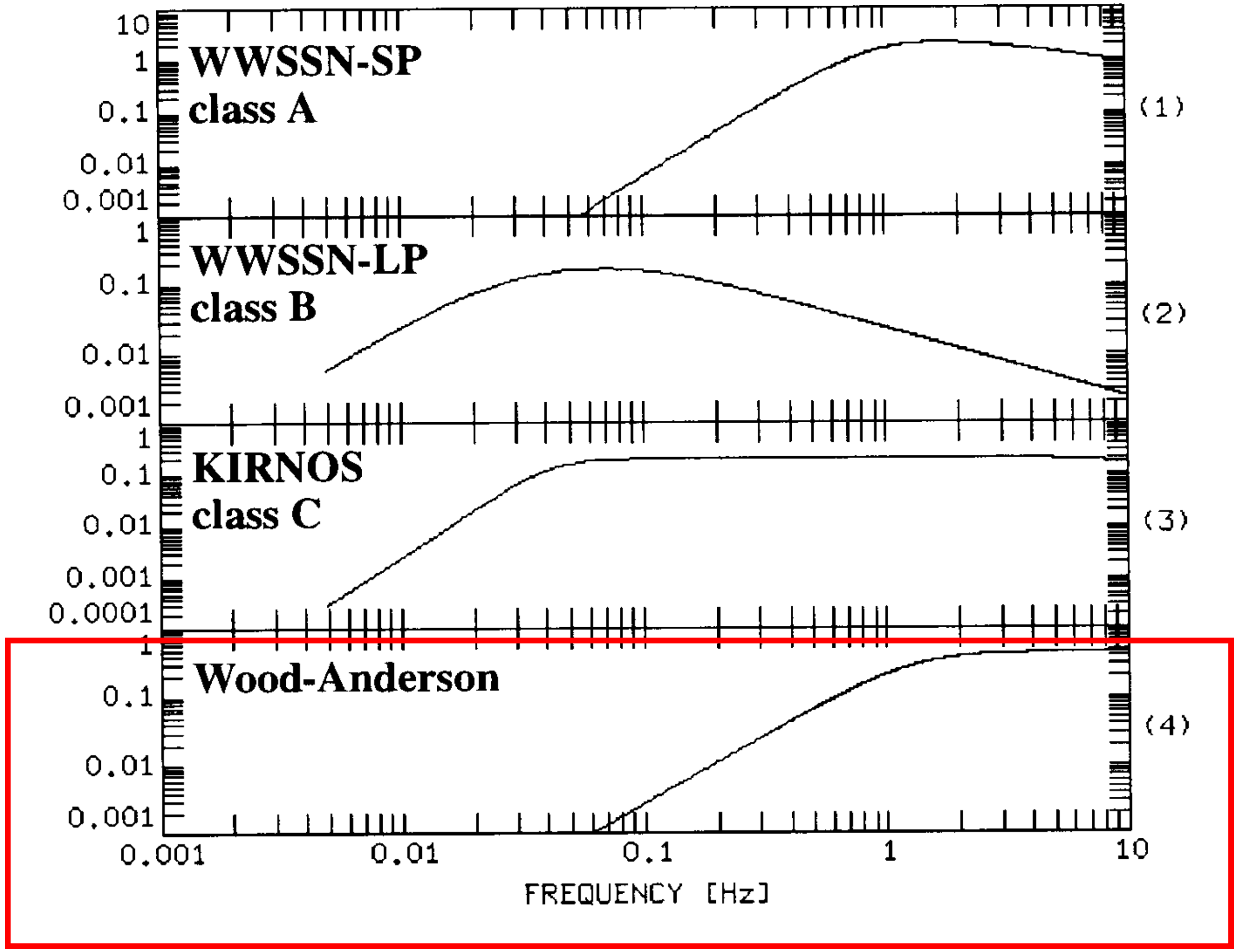


Dal nomogramma possiamo facilmente capire che il valore di magnitudo ricavata dipende dalla distanza della stazione e dall'ampiezza massima registrata.
Vedete che in maniera approssimativa invece della distanza possiamo utilizzare il tempo che intercorre tra l'arrivo S e l'arrivo P (che 'e funzione della distanza del terremoto).

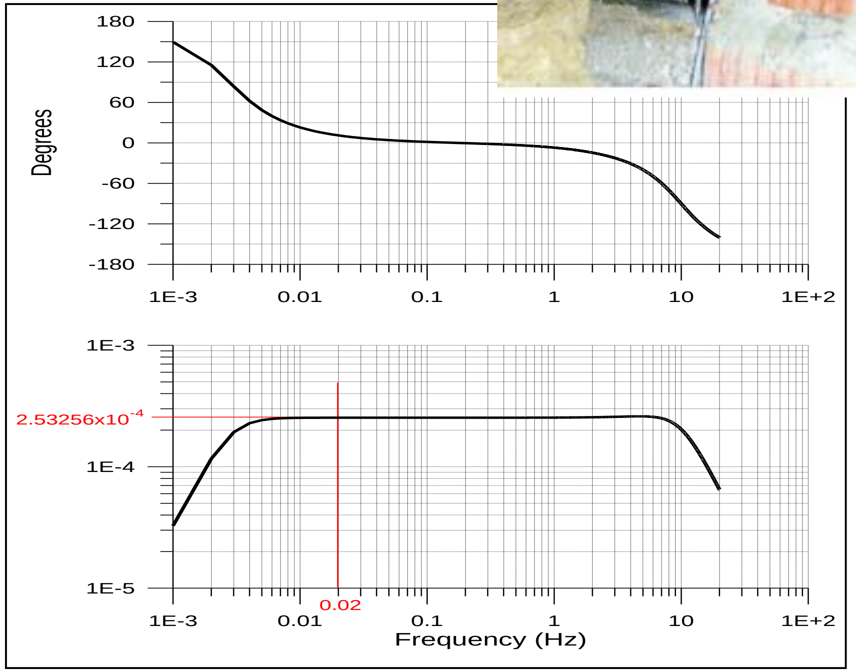
L'equazione per la Magnitudo Locale e':

ML = log10A*(mm)* + *(fattore correttivo per la distanza)*

Qui *A* e' l'ampiezza, in millimetri, misurata direttamente dalla registrazione. Richter defini che un terremoto ha magnitudo pari a 3 quando un evento registrato a 100 km di distanza con un sismometro di tipo Wood-Anderson con periodo proprio di 0.7 secondi (e non entriamo nel dettaglio della sismomentria per capire cosa significhi periodo proprio di un sismometro...) e 2800 ingrandimenti da luogo ad una ampiezza massima di 1 millimetro. Notate che il valore della magnitudo dipende dal logaritmo in base 10 dell'ampiezza. Quindi tra un terremoto di magnitudo 4 ed un terremoto di magnitudo 5 l'ampiezza varia di 10 volte!!



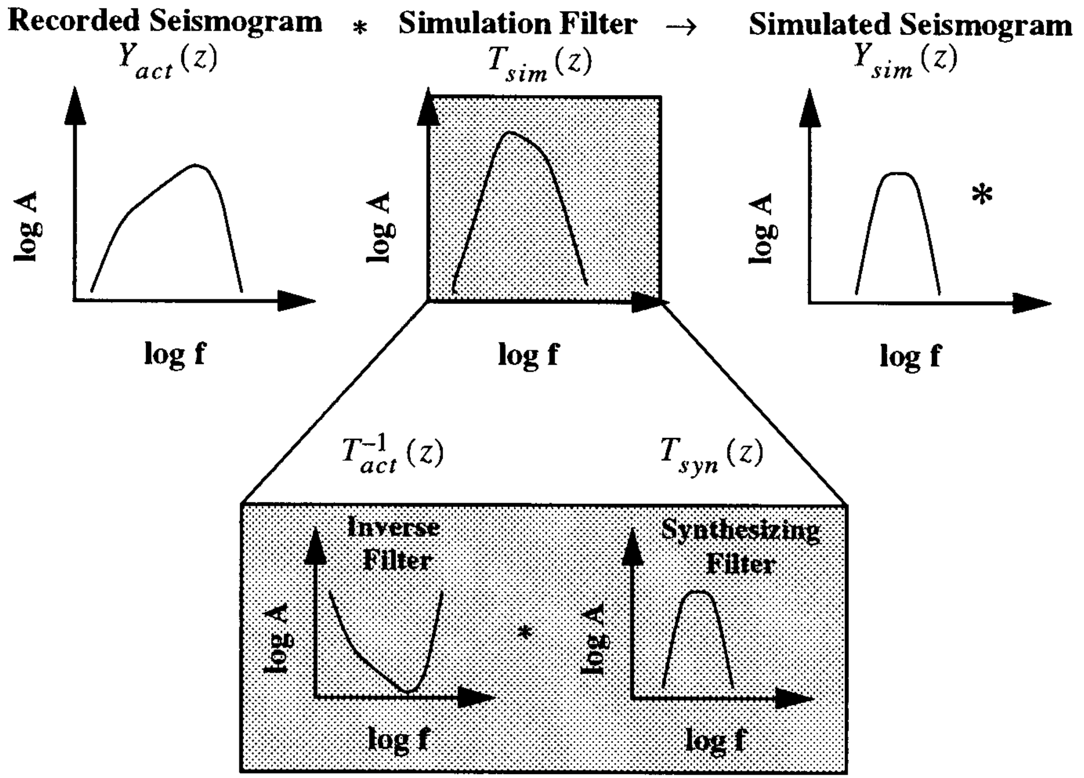
WWSSN-SP
class A
WWSSN-LP
class B
KIRNOS
class C
Wood-Anderson
(1)
(2)
(3)
(4)
10
1
0.1
0.01
0.001
0.0001
0.001
0.01
0.1
1
10
FREQUENCY [Hz]


Degrees
180
120
60
0
-60
-120
-180
1E-3
0.01
0.1
1
10
1E+2
1E-3
2.53256x10^{-4}
1E-4
1E-5
0.02
Frequency (Hz)

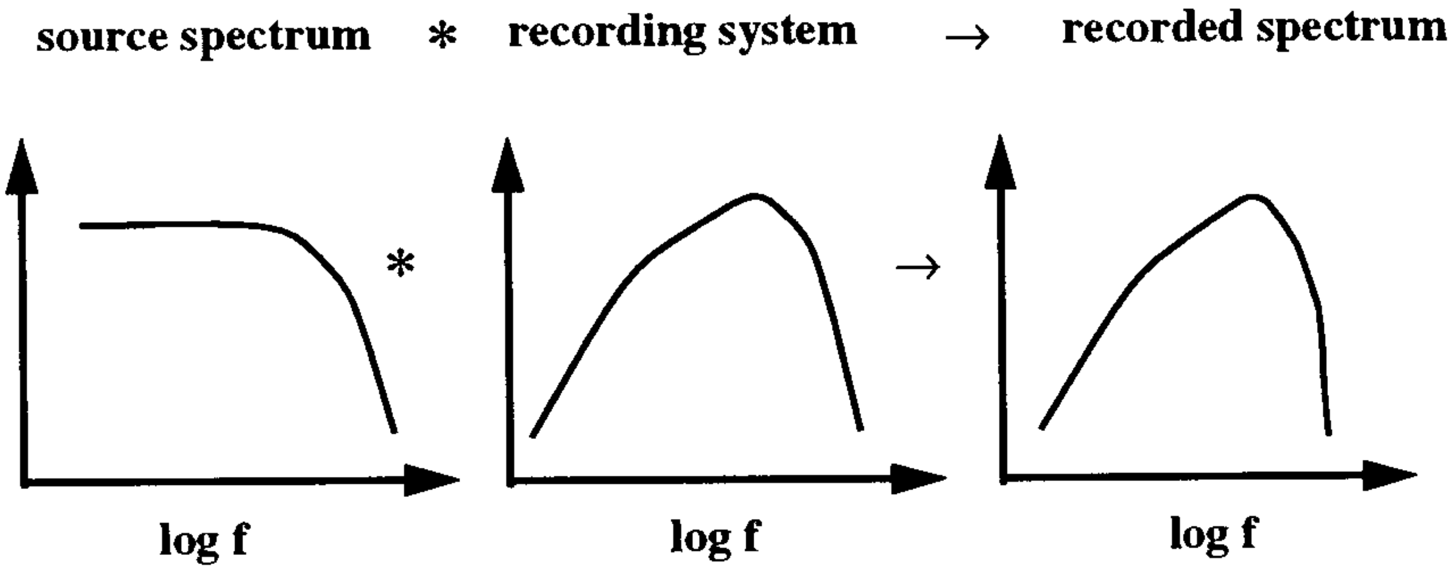
Per simulare un determinato sistema sismografico, dobbiamo prima rimuovere gli effetti del sistema di registrazione originale. Questo viene fatto filtrando i record digitali di un dato sismografo sistema con un filtro che approssima la risposta inversa del sistema di registrazione effettivo.
Quindi le serie temporali risultanti devono essere filtrate con una funzione di risposta che simula gli effetti dello strumento desiderato. Se $T_{act}(z)$ e $T_{sym}(z)$ sono la funzione di trasferimento dell'attuale strumento di registrazione e dello strumento da sintetizzare, e $Y_{act}(z)$ e $Y_{sim}(z)$ sono le trasformate z dei sismogrammi registrati e simulati, rispettivamente, la procedura di simulazione può essere descritta come segue:

$$Y_{sim}(z) = \frac{T_{syn}(z)}{T_{act}(z)} \cdot Y_{act}(z) = T_{sim}(z) \cdot Y_{act}(z) \quad . \tag{9.1}$$

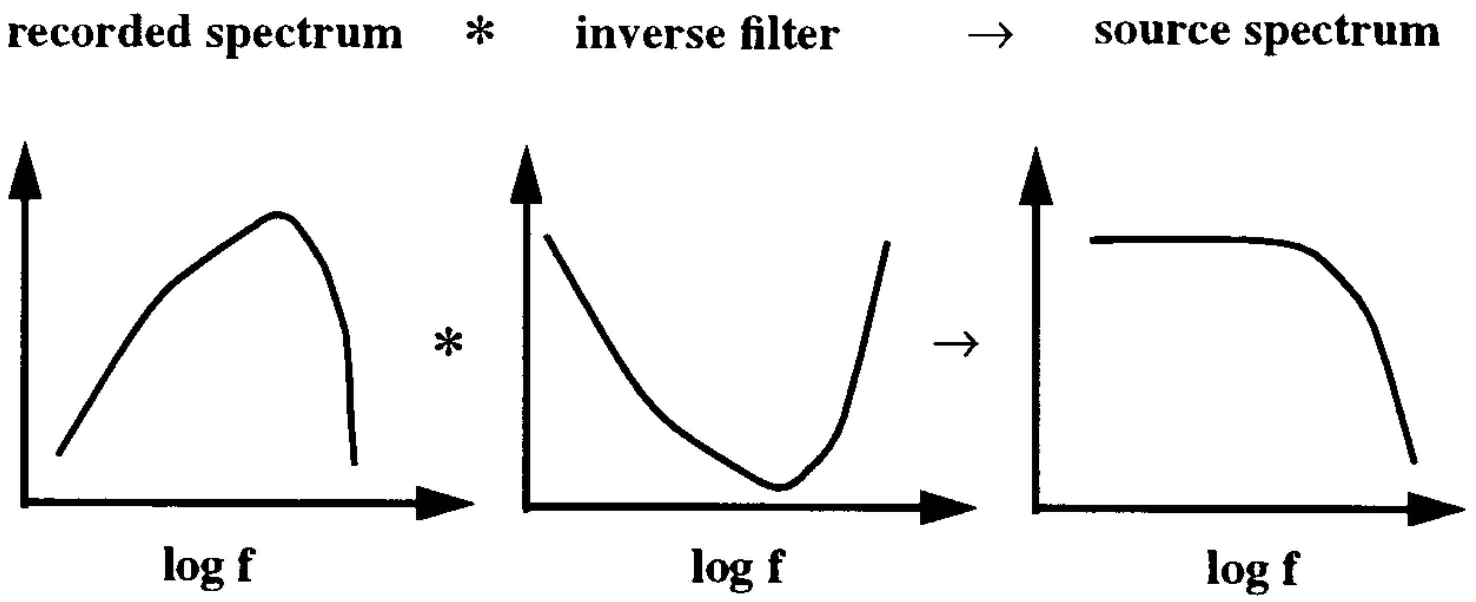
$$Y_{sim}(z) = \frac{T_{syn}(z)}{T_{act}(z)} \cdot Y_{act}(z) = T_{sim}(z) \cdot Y_{act}(z)$$



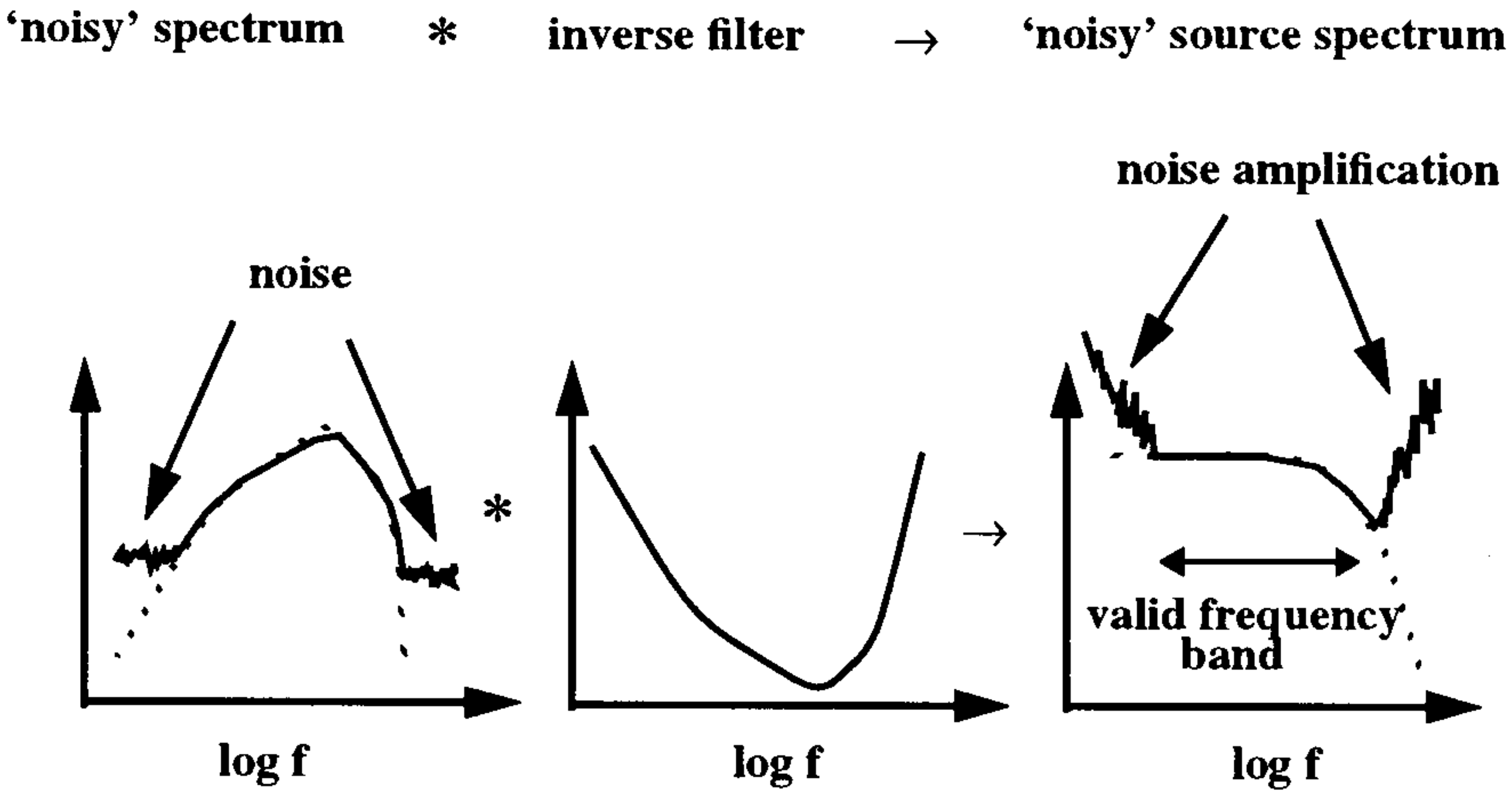
La simulazione di sismografi digitali. Il filtro di simulazione può essere pensato come una combinazione di un filtro inverso per il sistema di registrazione utilizzato e un filtro di sintesi per il sistema di registrazione simulato. Sono mostrato schizzi schematici delle funzioni di risposta in frequenza di ampiezza dei sottosistemi che contribuiscono.



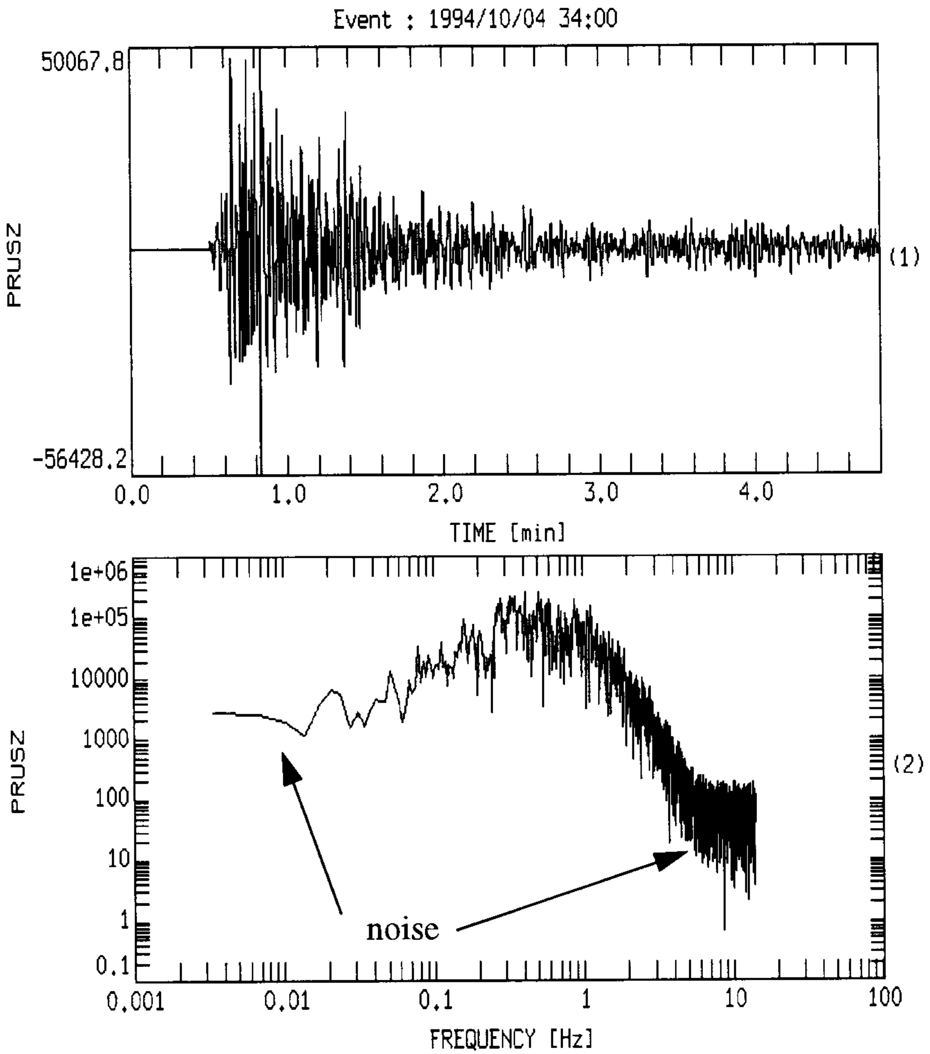
Registrazione dello spettro di spostamento di una sorgente di terremoto idealizzata.

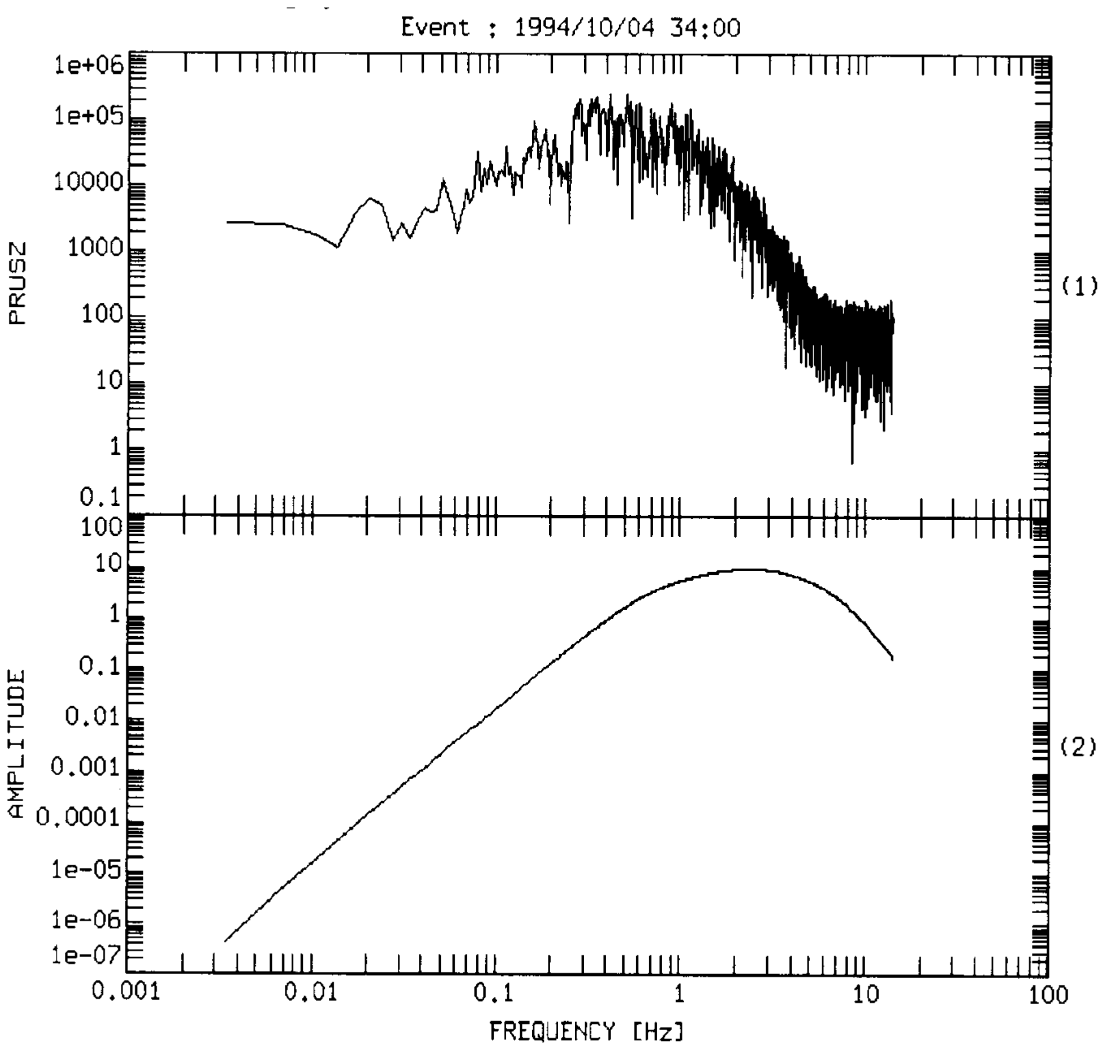


Ripristino dello spettro sorgente mediante filtro inverso nel caso privo di rumore

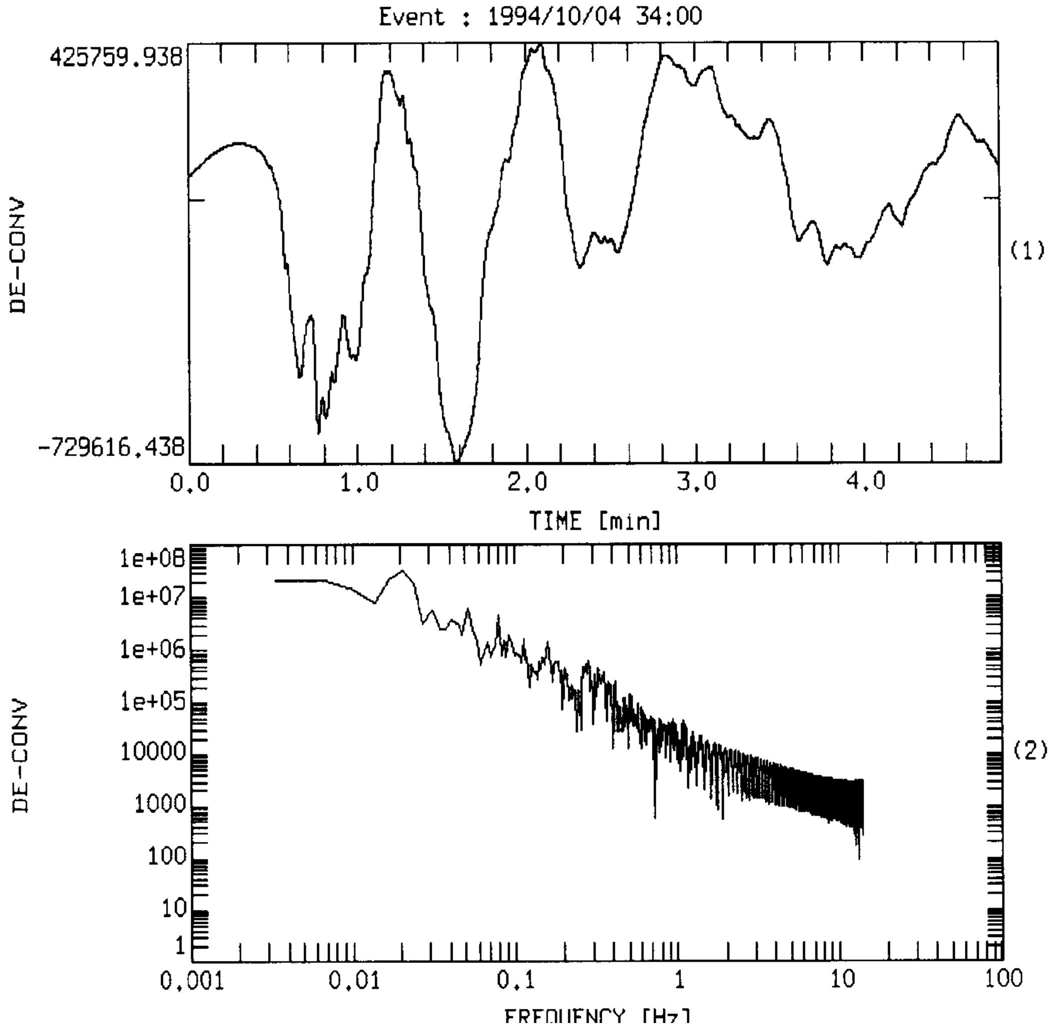


Amplificazione del rumore mediante filtro inverso. La linea continua nel pannello di sinistra mostra il segnale più il rumore mentre il segnale privo di rumore è mostrato dalla linea tratteggiata.

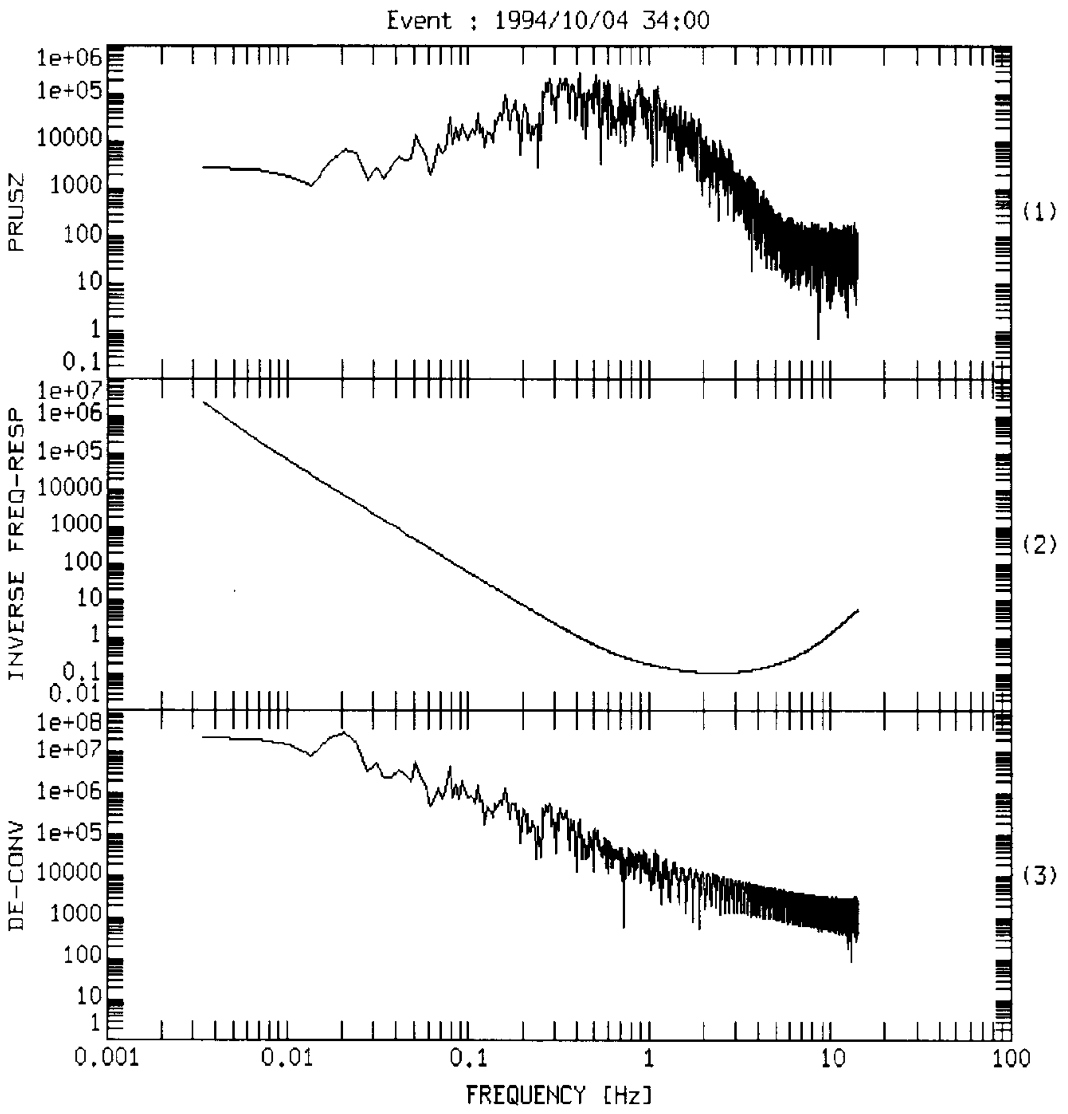

Event : 1994/10/04 34:00
50067.8
-56428.2
PRUSZ
(1)
0.0
1.0
2.0
3.0
4.0
TIME [min]
1e+06
1e+05
10000
1000
100
10
1
0.1
PRUSZ
(2)
noise
0.001
0.01
0.1
1
10
100
FREQUENCY [Hz]



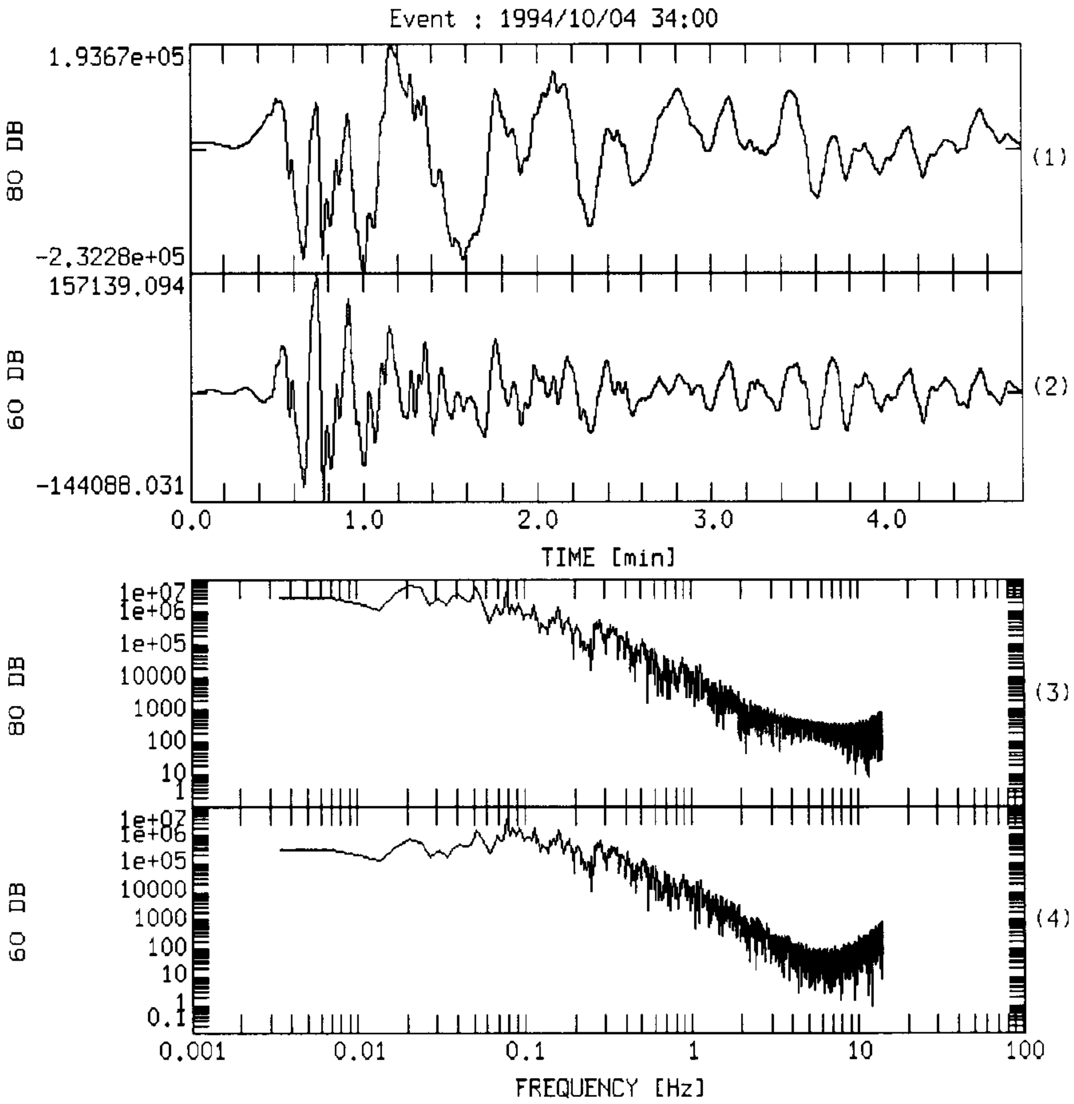
Event : 1994/10/04 34:00
PRUSZ
1e+06
1e+05
10000
1000
100
10
1
0.1
(1)
AMPLITUDE
100
10
1
0.1
0.01
0.001
0.0001
1e-05
1e-06
1e-07
(2)
0.001
0.01
0.1
1
10
100
FREQUENCY [Hz]

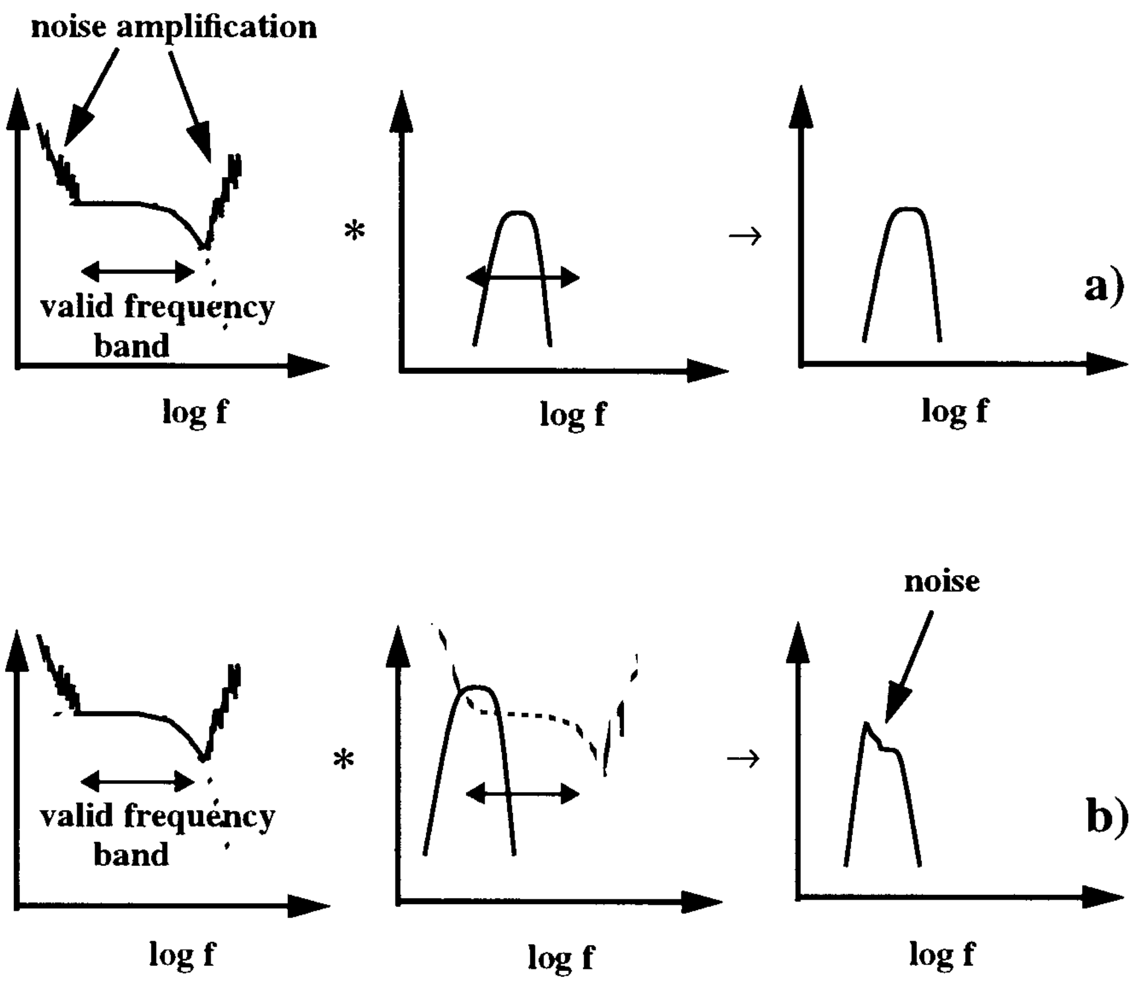


Event : 1994/10/04 34:00
425759.938
-729616.438
DE-CONV
(1)
0.0
1.0
2.0
3.0
4.0
TIME [min]
1e+08
1e+07
1e+06
1e+05
10000
1000
100
10
1
DE-CONV
(2)
0.001
0.01
0.1
1
10
100
FREQUENCY [Hz]

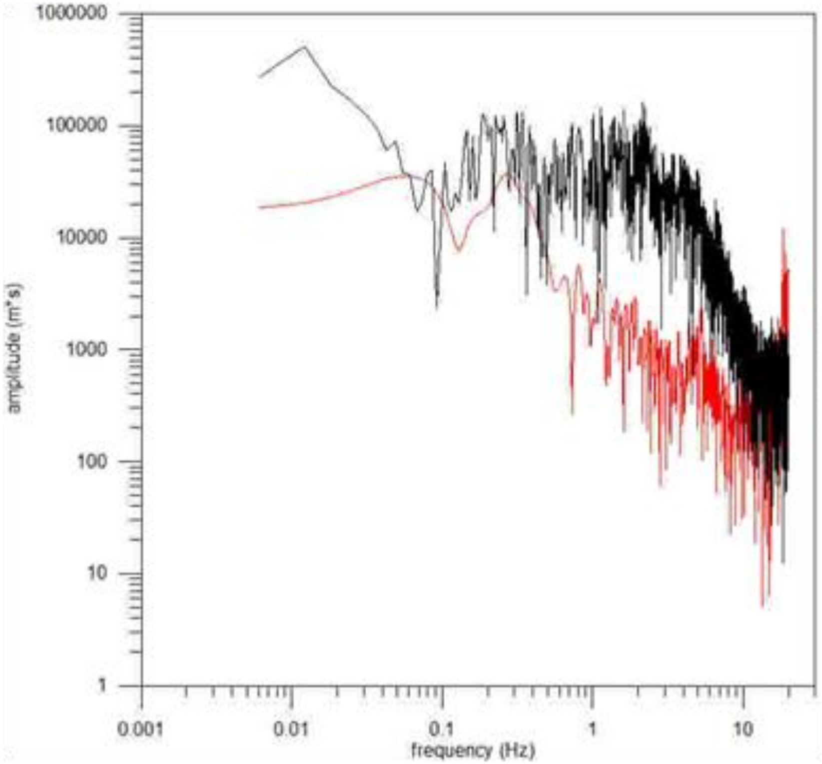


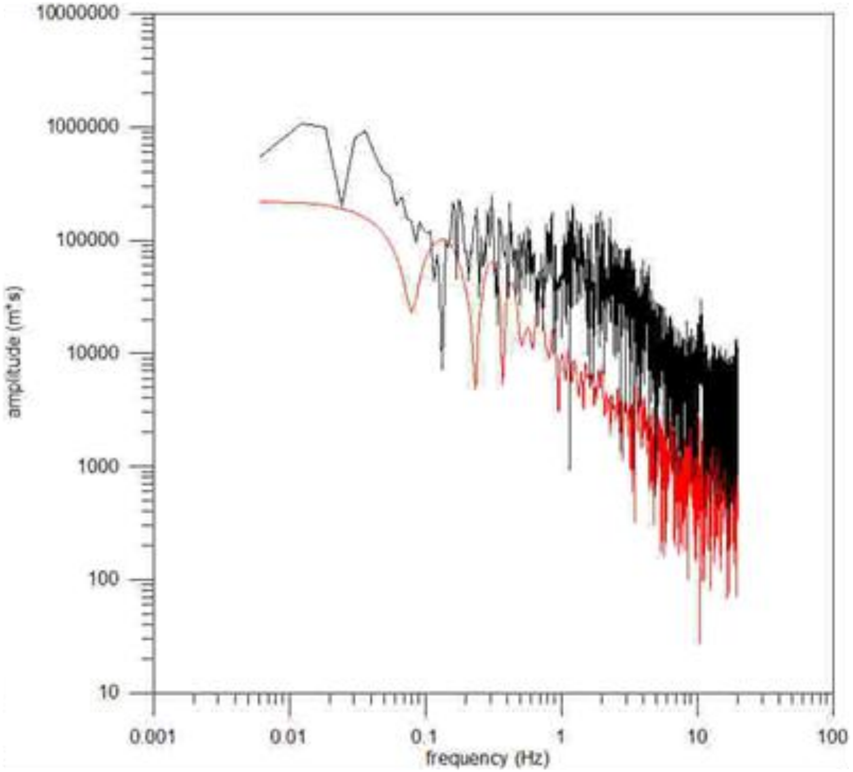
Event : 1994/10/04 34:00
PRUSZ
INVERSE FREQ-RESP
DE-CONV
(1)
(2)
(3)
FREQUENCY [Hz]
0.001
0.01
0.1
1
10
100


Event : 1994/10/04 34:00
1.9367e+05
-2.3228e+05
80 DB
(1)
157139.094
-144088.031
60 DB
(2)
0.0
1.0
2.0
3.0
4.0
TIME [min]
1e+07
1e+06
1e+05
10000
1000
100
10
1
80 DB
(3)
1e+07
1e+06
1e+05
10000
1000
100
10
1
0.1
60 DB
(4)
0.001
0.01
0.1
1
10
100
FREQUENCY [Hz]

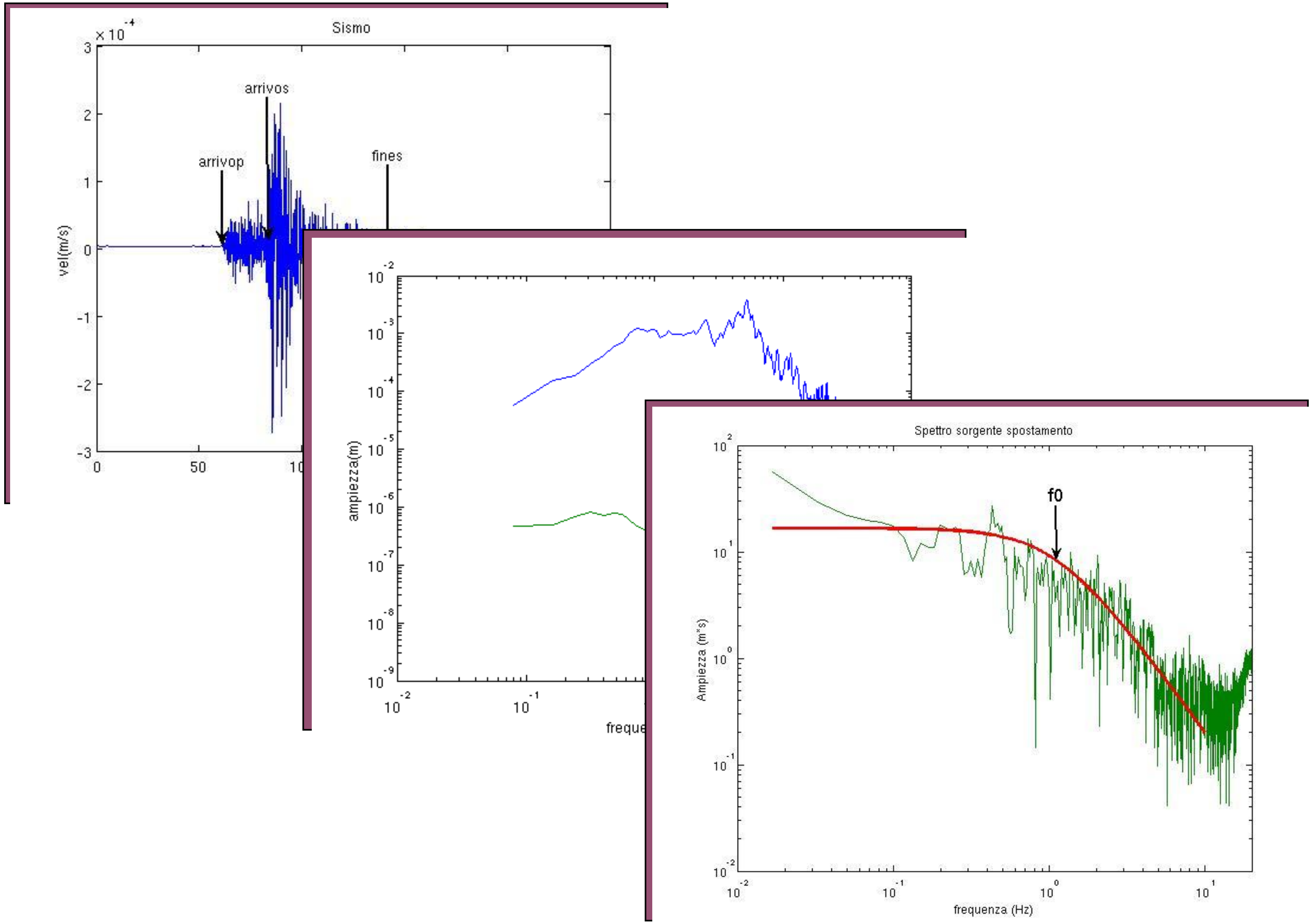


Mappatura del rumore di deconvoluzione nel sistema simulato. a) La banda passante dello strumento simulato rientra pienamente nella banda di frequenza valida. b) La banda passante dello strumento simulato va parte al di fuori della banda di frequenza valida. Da qui il rumore di deconvoluzione viene mappato nel sistema simulato.


1000000
100000
10000
1000
100
10
1
amplitude (m*s)
0.001
0.01
0.1
1
10
frequency (Hz)


10000000
1000000
100000
10000
1000
100
10
amplitude (m*s)
0.001
0.01
0.1
1
10
100
frequency (Hz)

Implementazione del calcolo della magnitudo momento nel sistema real-time



Trigger

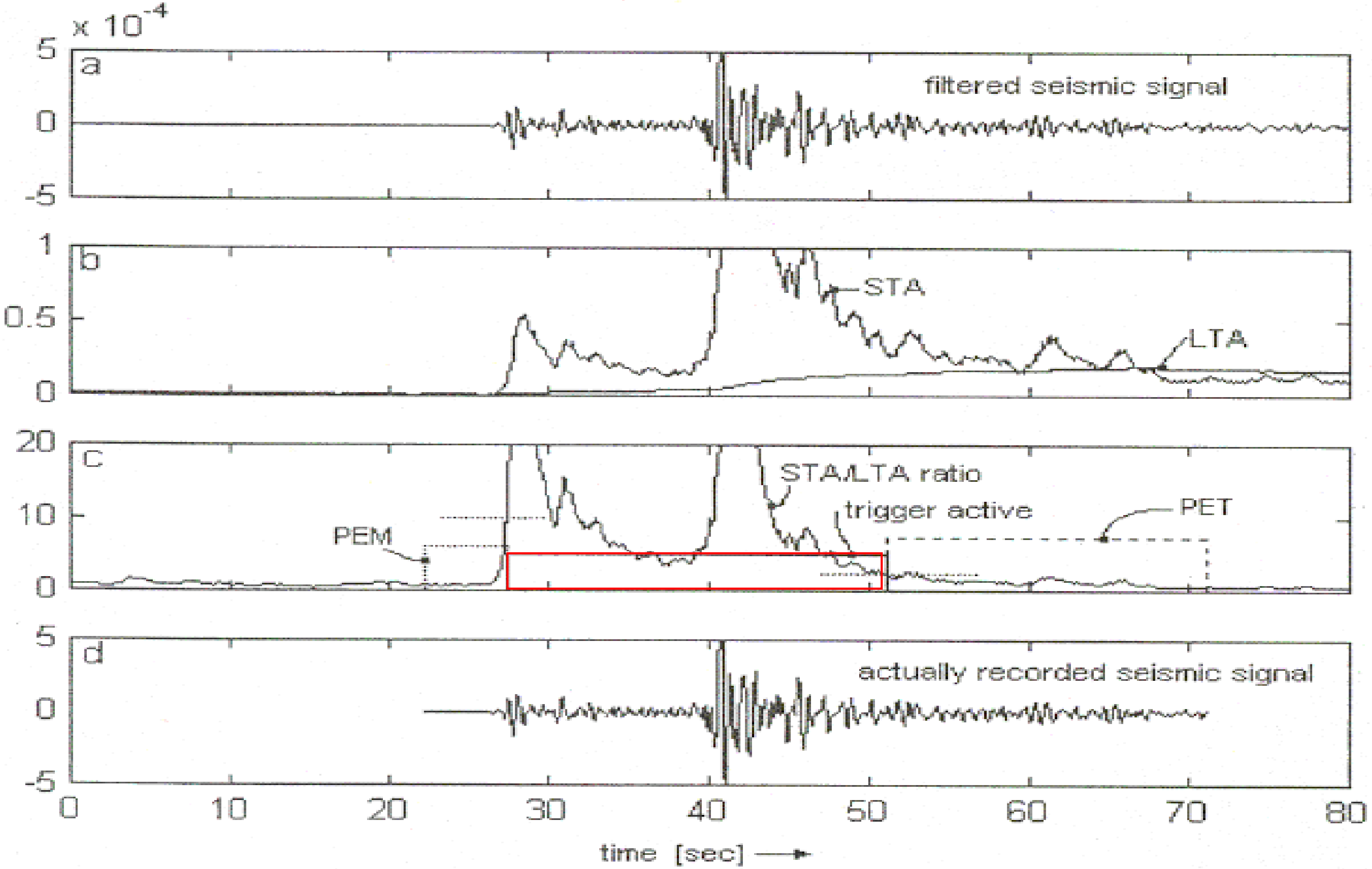
- **<u>A soglia</u>**: il trigger viene attivato quando l’ampiezza del segnale supera una predefinita soglia

- **<u>RMS</u>** (root-mean-square): viene considerata l’ampiezza del segnale in una breve finestra temporale al posto dell’ampiezza istantanea.

- **<u>STA/LTA</u>**: viene considerato il rapporto della media del segnale relativo ad una finestra temporale corta (STA, short time average) rispetto ad una finestra temporale lunga (LTA, long time average).

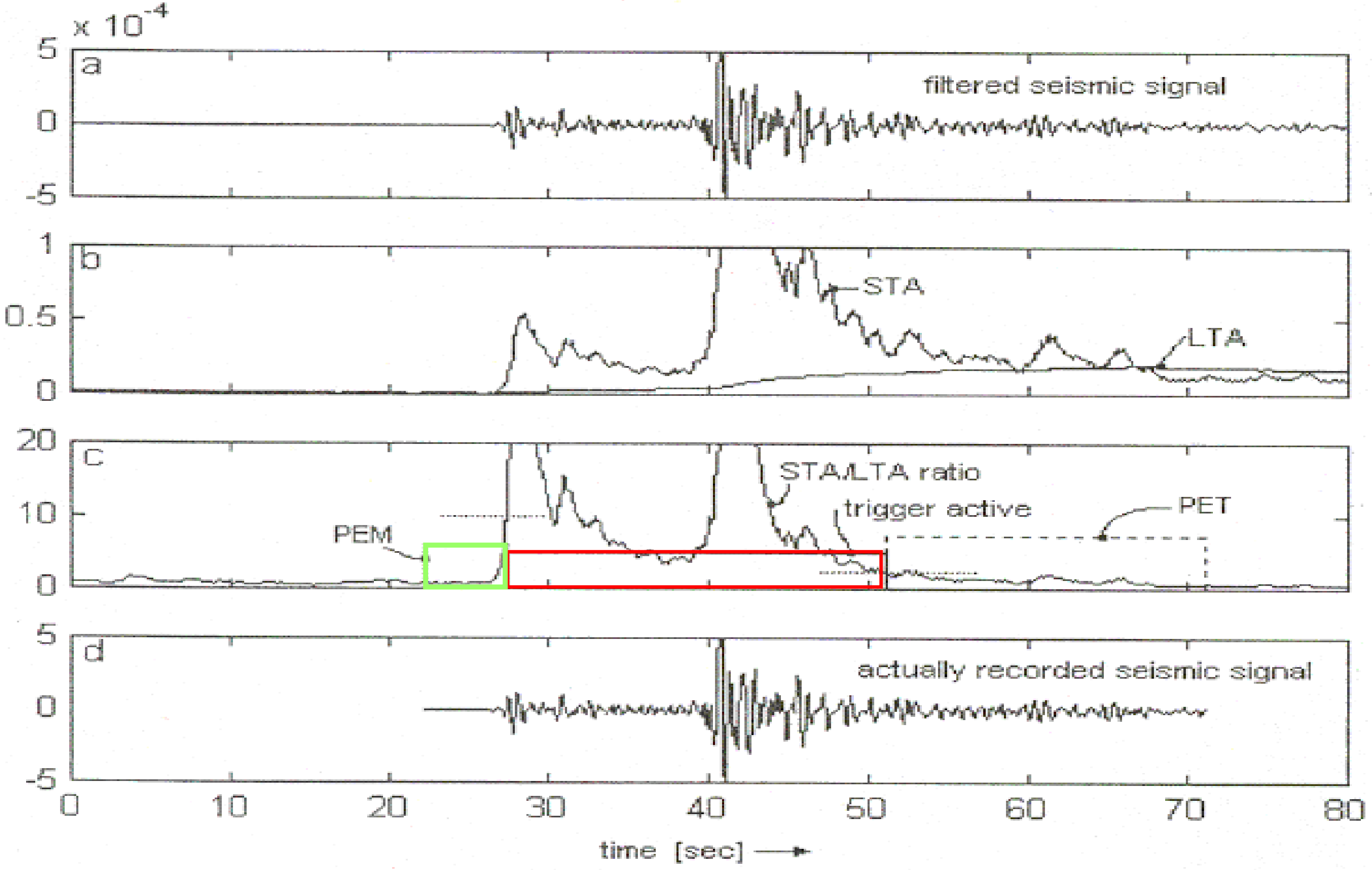
STA – short time average **PEM – Pre-event time**

LTA – long time average **PET - post event time**



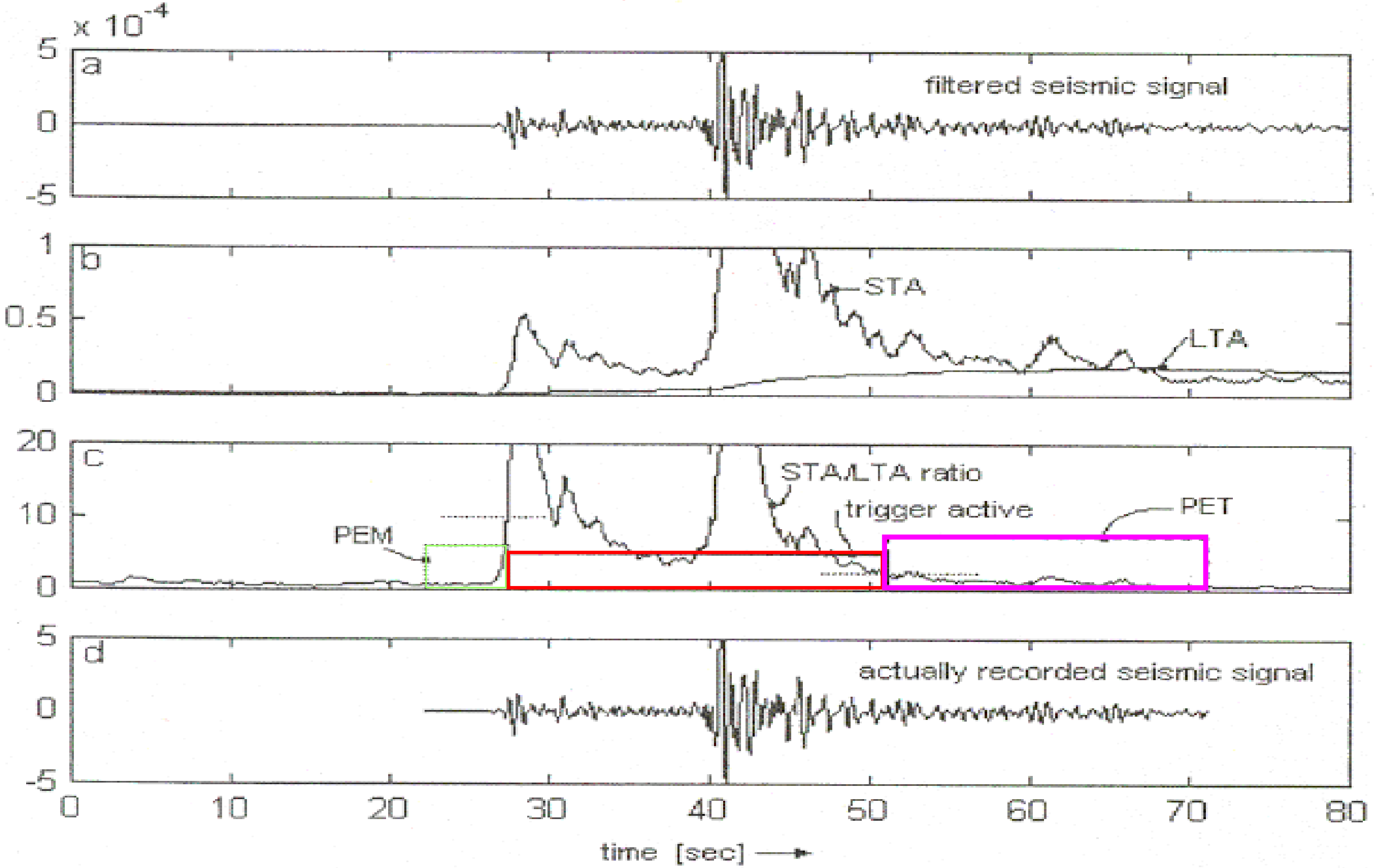
STA – short time average
LTA – long time average

PEM – Pre-event time
PET - post event time



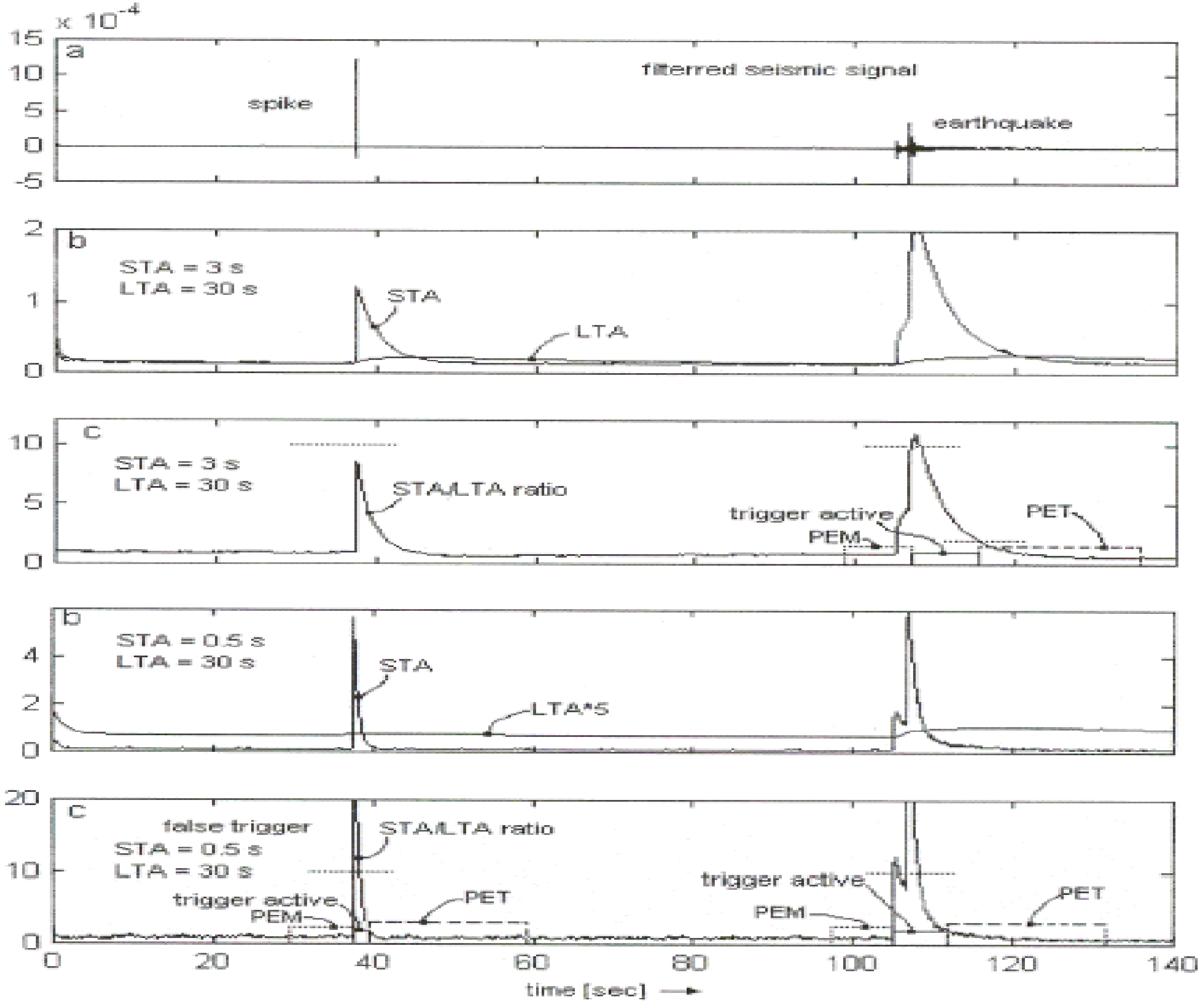
STA – short time average
LTA – long time average

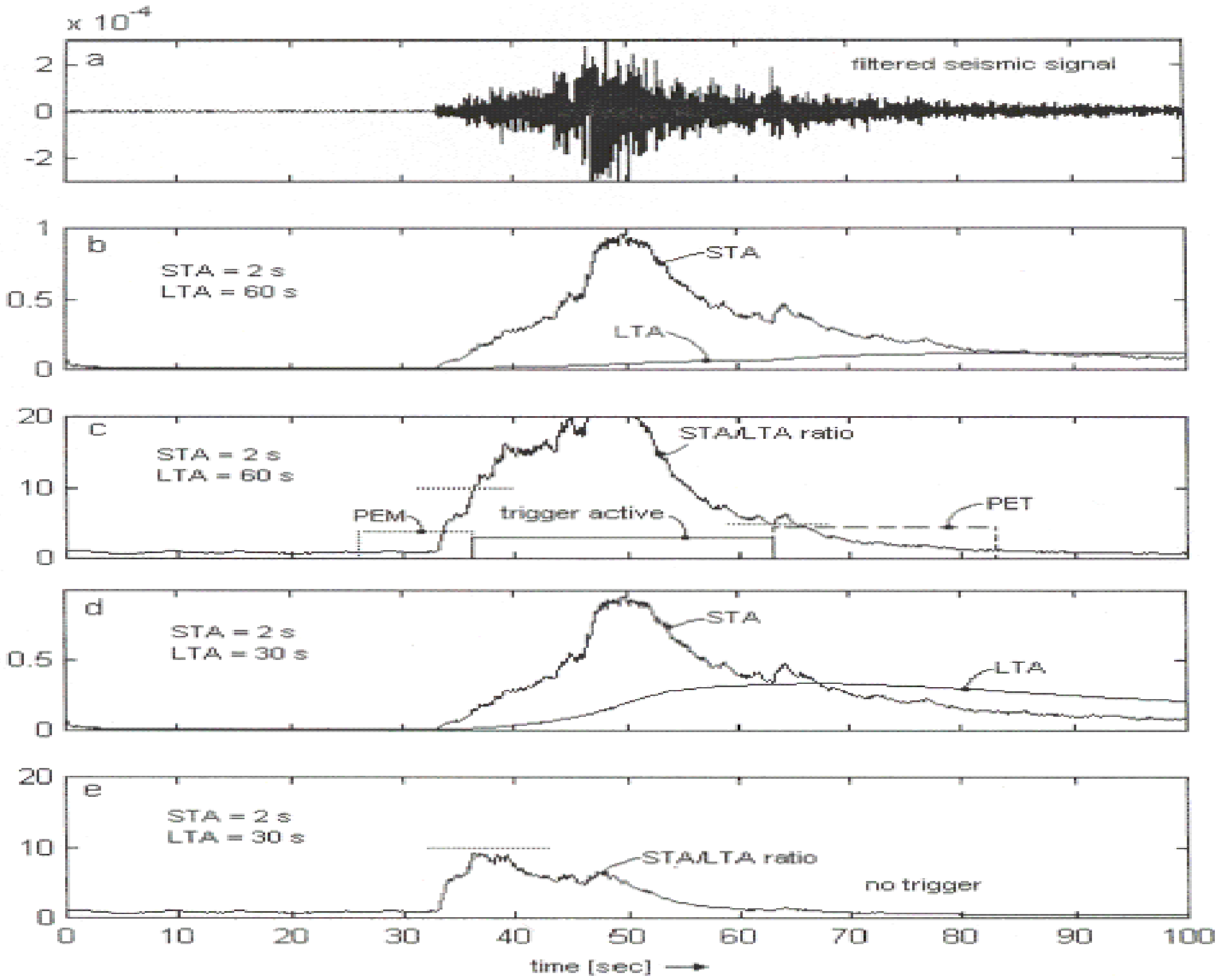
PEM – Pre-event time
PET - post event time



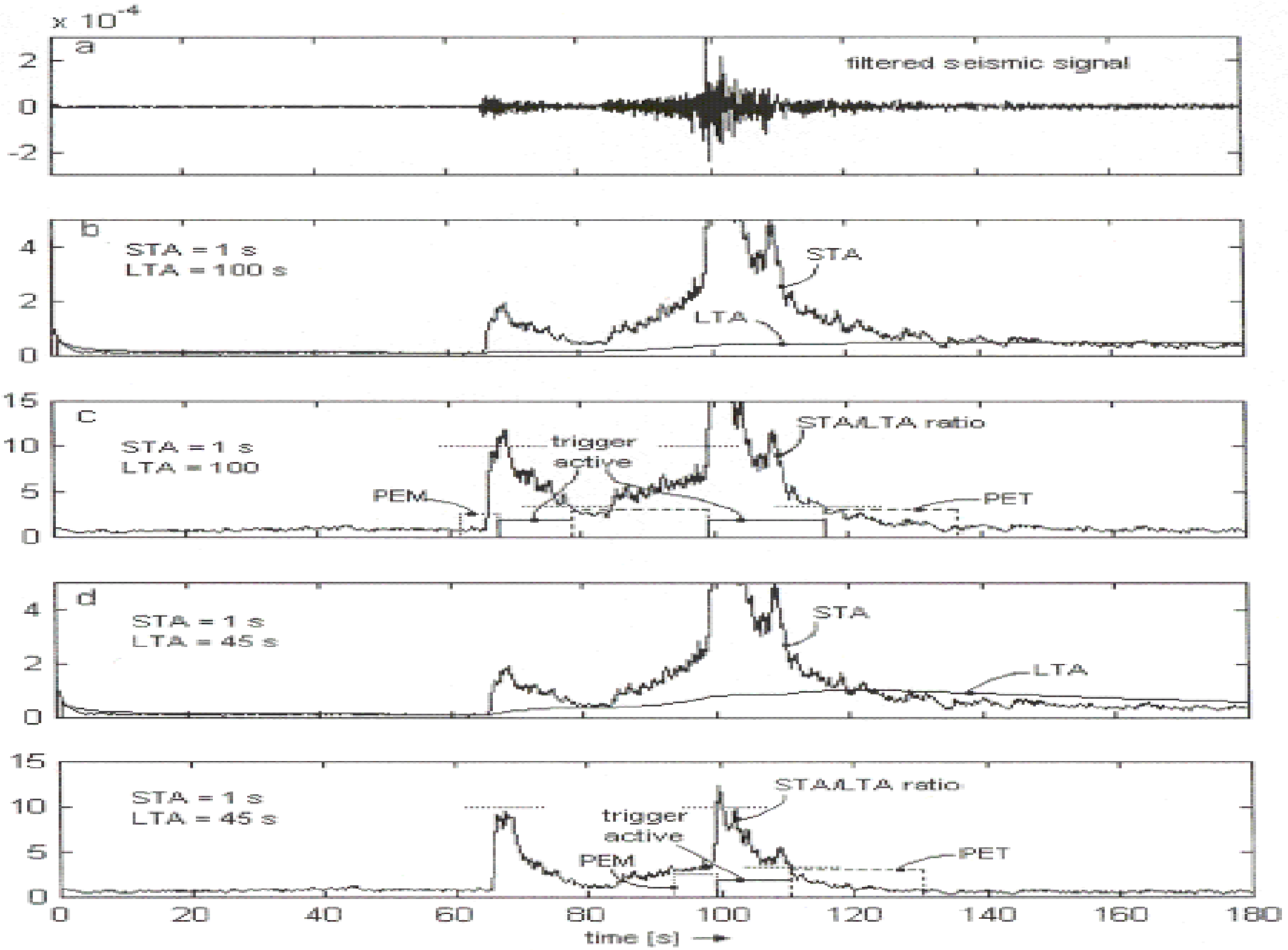
- durata finestra STA
- durata finestra LTA
- valore della soglia di trigger del rapporto STA/LTA
- valore della soglia di detrigger del rapporto STA/LTA

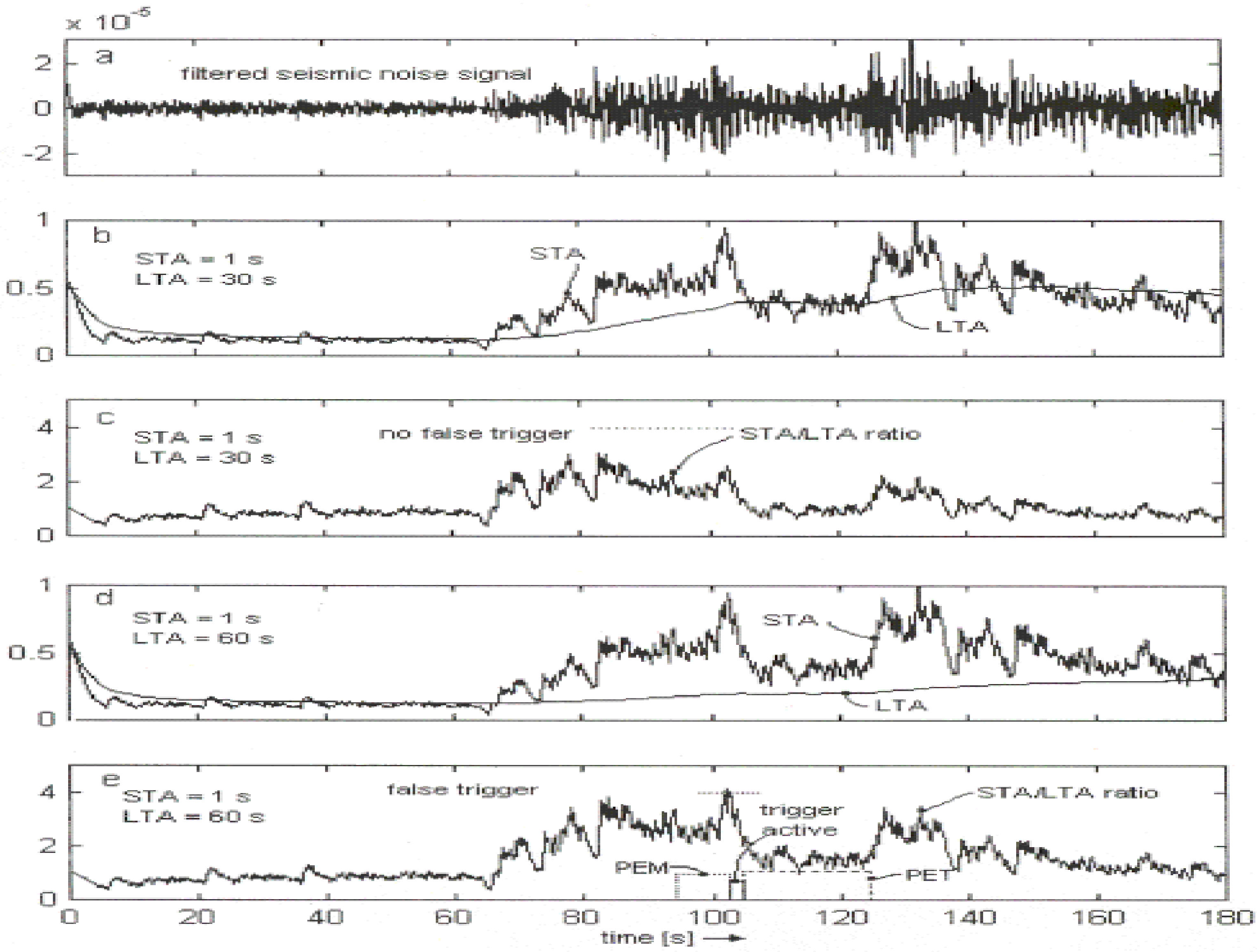
- filtri del trigger
- durata della finestra pre-evento (PEM)
- durata della finestra post-evento (PET)
- schema della dichiarazione del trigger

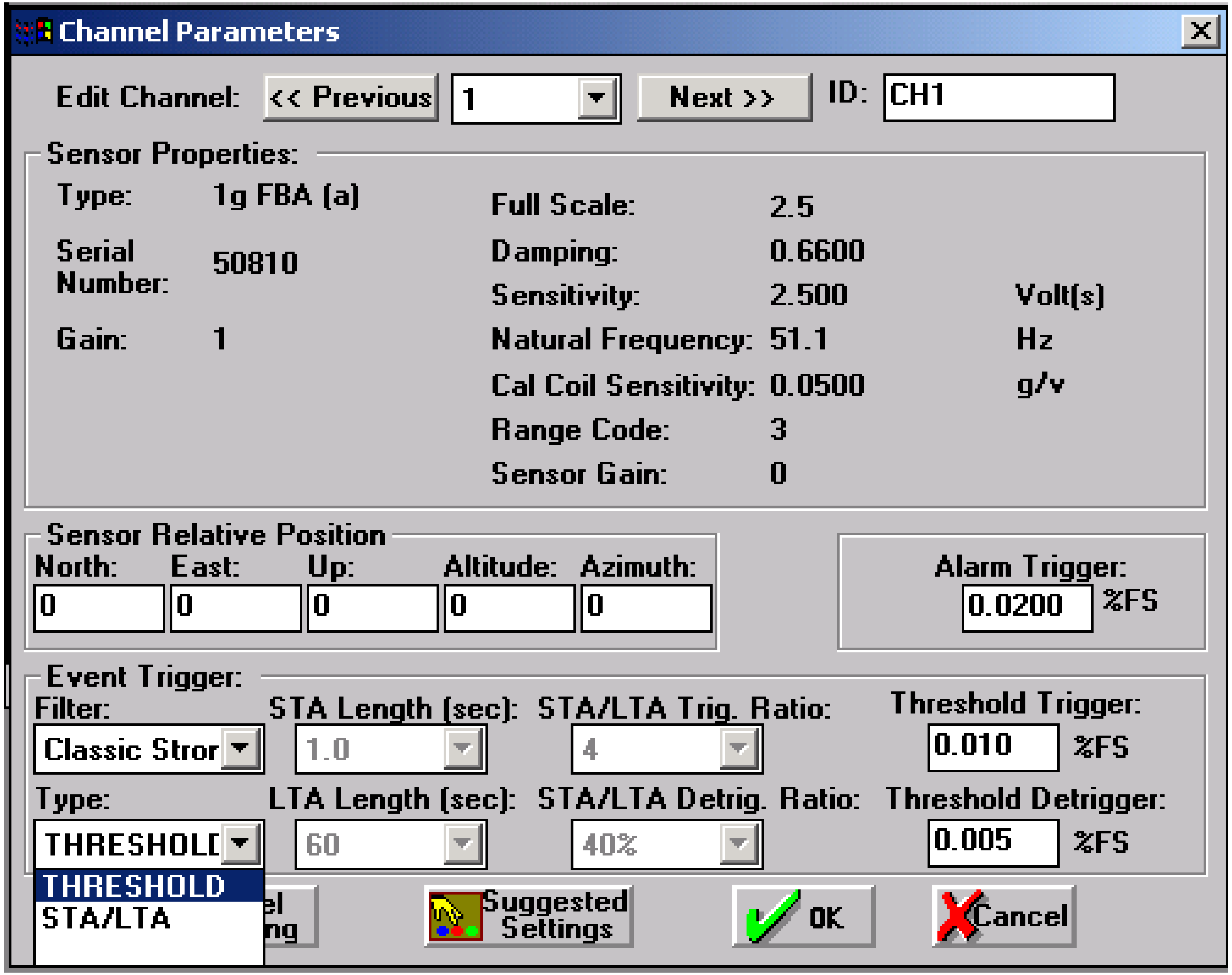

x 10^-4
a
spike
filterred seismic signal
earthquake
b
STA = 3 s
LTA = 30 s
STA
LTA
c
STA = 3 s
LTA = 30 s
STA/LTA ratio
trigger active
PEM
PET
b
STA = 0.5 s
LTA = 30 s
STA
LTA*5
c
false trigger
STA = 0.5 s
LTA = 30 s
STA/LTA ratio
trigger active
PEM
PET
trigger active
PEM
PET
time [sec]

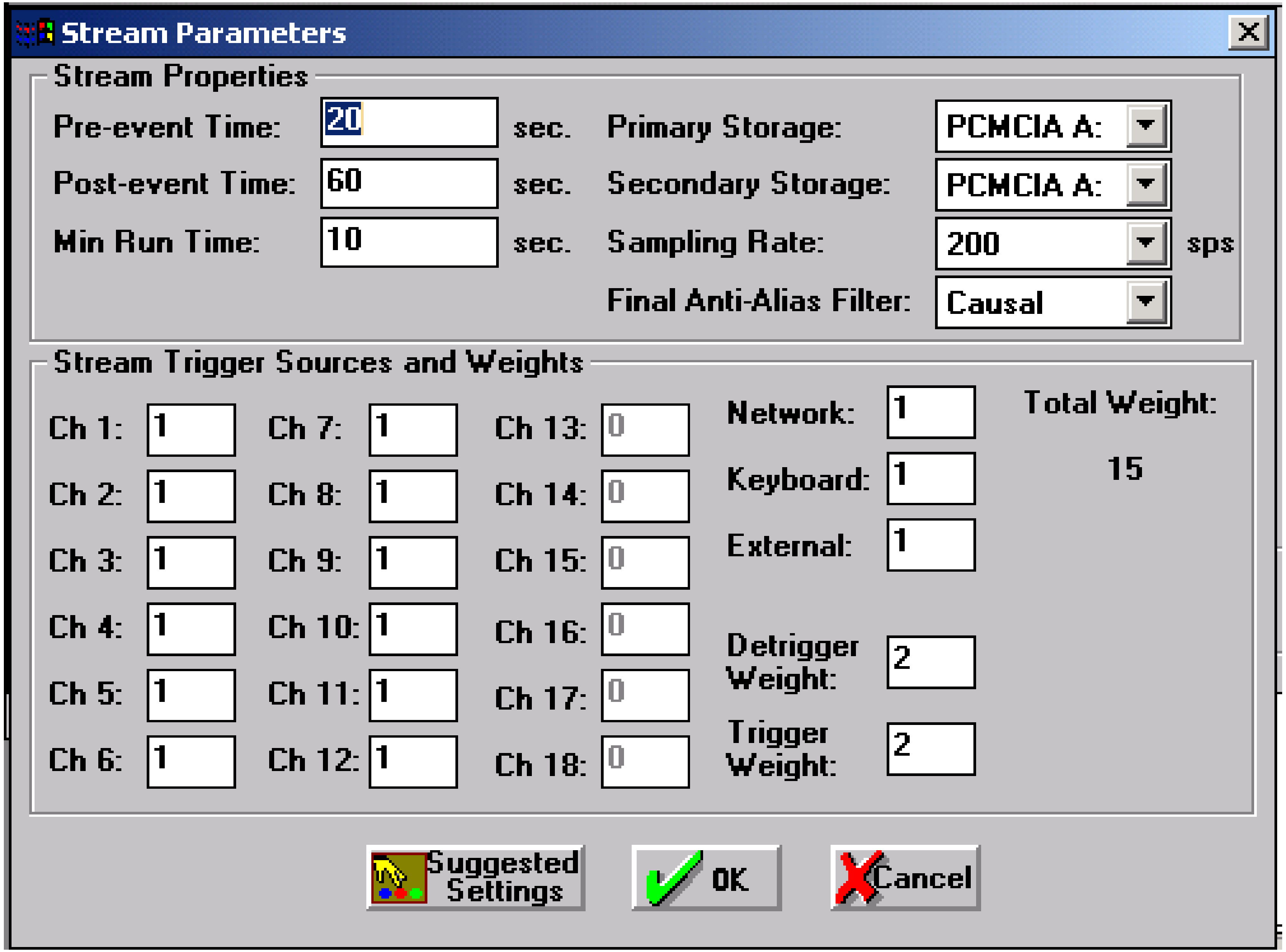


x 10^-4
a
filtered seismic signal
b
STA = 2 s
LTA = 60 s
STA
LTA
c
STA = 2 s
LTA = 60 s
STA/LTA ratio
PEM
trigger active
PET
d
STA = 2 s
LTA = 30 s
STA
LTA
e
STA = 2 s
LTA = 30 s
STA/LTA ratio
no trigger
time [sec]

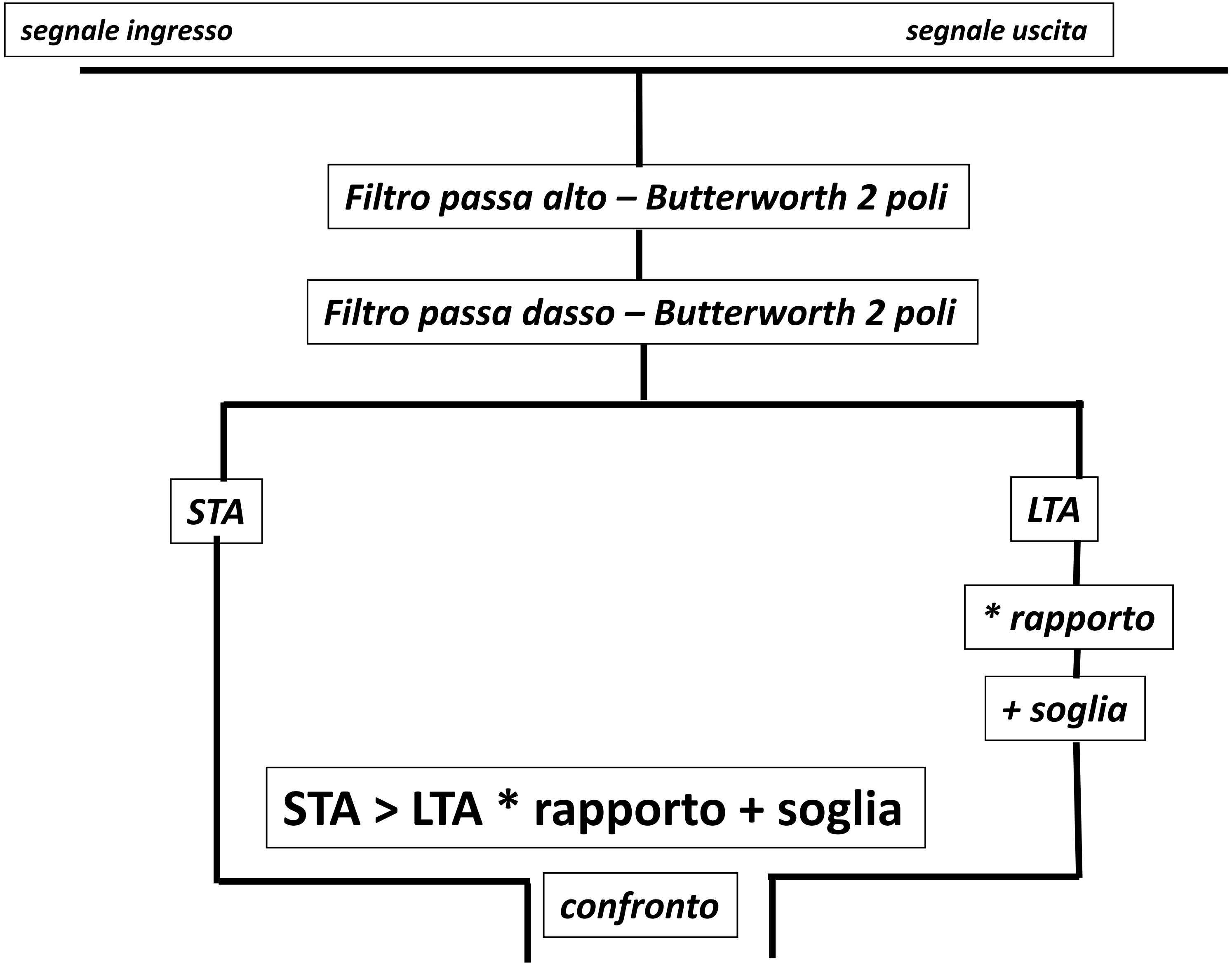

x 10^-4
a
filtered seismic signal
b
STA = 1 s
LTA = 100 s
STA
LTA
c
STA = 1 s
LTA = 100
STA/LTA ratio
trigger
active
PEM
PET
d
STA = 1 s
LTA = 45 s
STA
LTA
STA = 1 s
LTA = 45 s
STA/LTA ratio
trigger
active
PEM
PET
time [s]


x 10^-5
a
filtered seismic noise signal
b
STA = 1 s
LTA = 30 s
STA
LTA
c
STA = 1 s
LTA = 30 s
no false trigger
STA/LTA ratio
d
STA = 1 s
LTA = 60 s
STA
LTA
e
STA = 1 s
LTA = 60 s
false trigger
trigger active
STA/LTA ratio
PEM
PET
time [s]


Channel Parameters
Edit Channel:
<< Previous
1
Next >>
ID:
CH1
Sensor Properties:
Type: 1g FBA (a)
Serial Number: 50810
Gain: 1
Full Scale: 2.5
Damping: 0.6600
Sensitivity: 2.500 Volt(s)
Natural Frequency: 51.1 Hz
Cal Coil Sensitivity: 0.0500 g/v
Range Code: 3
Sensor Gain: 0
Sensor Relative Position
North: 0
East: 0
Up: 0
Altitude: 0
Azimuth: 0
Alarm Trigger:
0.0200 %FS
Event Trigger:
Filter:
Classic Stror
STA Length (sec):
1.0
STA/LTA Trig. Ratio:
4
Threshold Trigger:
0.010 %FS
Type:
THRESHOLD
LTA Length (sec):
60
STA/LTA Detrig. Ratio:
40%
Threshold Detrigger:
0.005 %FS
THRESHOLD
STA/LTA
Suggested Settings
OK
Cancel



Stream Parameters
Stream Properties
Pre-event Time: 20 sec.
Post-event Time: 60 sec.
Min Run Time: 10 sec.
Primary Storage: PCMCIA A:
Secondary Storage: PCMCIA A:
Sampling Rate: 200 sps
Final Anti-Alias Filter: Causal
Stream Trigger Sources and Weights
Ch 1: 1
Ch 2: 1
Ch 3: 1
Ch 4: 1
Ch 5: 1
Ch 6: 1
Ch 7: 1
Ch 8: 1
Ch 9: 1
Ch 10: 1
Ch 11: 1
Ch 12: 1
Ch 13: 0
Ch 14: 0
Ch 15: 0
Ch 16: 0
Ch 17: 0
Ch 18: 0
Network: 1
Keyboard: 1
External: 1
Detrigger Weight: 2
Trigger Weight: 2
Total Weight:
15
Suggested Settings
OK
Cancel


segnale ingresso
segnale uscita
Filtro passa alto – Butterworth 2 poli
Filtro passa dasso – Butterworth 2 poli
STA
LTA
* rapporto
+ soglia
STA > LTA * rapporto + soglia
confronto