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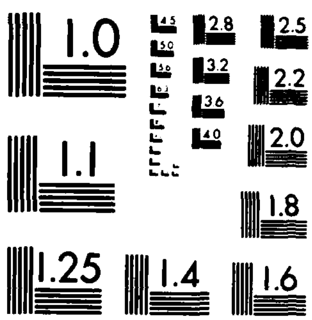

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TECHNICAL REPORT NO. 385

A PHASE CORRECTION FILTER FOR WAVEFORM RECOVERY OF COMPRESSED DIRECT-RECORDED OCEAN BOTTOM SEISMOMETER DATA

by

Ules S Wade
Clive R.B. Lister

Office of Naval Research
Contract N-00014-75-C-0502
MOD P00021
Project NR 083-012

Reference M81-05
March 1981

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A phase correction filter for waveform recovery of compressed direct-recorded ocean bottom seismometer data.

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C.R.B. Lister
Principal Investigator

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George Anderson
Associate Chairman for Research

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ABSTRACT

A phase convolution filter has been developed for the University of Washington's ocean bottom seismometers (OBS) to correct severe waveform distortion on playback. Signals are recorded directly on magnetic tape after compression by taking the bipolar square-root of the signal. Prior to digitization, the analog playback head shifts the phases of the low frequency harmonics and thus distorts the recorded waveform. If the distorted signal is squared without correction, the original waveforms and bandwidths are not recovered, leading to errors in the selection of arrival times and an inability to analyse the seismic coda for late arrivals. The phase-correcting filter has been constructed by inverting a smoothed version of the spike spectrum that results when a square wave is recorded on the tape.

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INTRODUCTION

Ocean bottom seismometers have been used in marine geophysical investigations to eliminate the shortcomings of free floating surface receivers. The University of Washington's OBS as described by Lister and Lewis, 1976, was designed with simplicity as the primary goal and has had a very high recovery rate both for the instruments and for usable data (Johnson et al, 1977). We will describe here only those features of the OBS pertinent to the solution of the waveform fidelity problem that arose when the direct-recorded data tapes were played back into a digitizer.

Three orthogonal components of motion are measured using L 15 B Mark Products 4.5 hertz geophones. They are leveled in a boat floating on silicone oil of a thickness chosen to place the high pass filter corner frequency well below 0.5 hertz. The signal from the sensors is amplified and passed through two three-pole Bessel filters to define a 2 to 100 hertz passband with a minimum of time-envelope distortion. The filtering limits the signal to within the band pass capabilities of the tape recorder. Dynamic-range compression is accomplished by taking the square root of the signal deviation, negative values being handled by a separate circuit, and the results recombined with sign preservation. Square-rooting generates low harmonic and cross-modulation distortion, consistent with substantial compression, and does not require a calibration level, since the transition to linear response can be made low enough to vanish in the background noise level. The harmonic content of a square-rooted sine wave is about 26 percent, and is independent of amplitude. The signal is recorded by a slow speed direct method on tape moving at 1mm per second, together with a clock signal encoded with the day, hour, minute, and second. On playback, a mini-computer is used to digitize all four channels and decode the clock in software. The analog tapes are played back on a slightly modified Hewlett-Packard 3960 series tape recorder, at a speed approximately 25 times the recording speed. A four-pole 20 hertz notch filter is used to eliminate the noise induced by vibration of the OBS tape drive motor. The signal is amplified by a tracking three-channel buffer amplifer until the largest signal peaks approach the limits of the digitizer's range. The sample spacing is 0.01 seconds in OBS time, or 2500 samples per second at playback speed, so that a 1250 hertz low-pass Butterworth filter could be inserted in the analog signal processing chain to reduce aliasing.

In figure 1, the measured frequency response of the electronic package (filled circles) is compared to the calculated response (open circles) of the electronics plus geophone sensor (Lee Bond, unpublished technical report). A 2 volt signal was inserted at the calibration input, attenuated by 1000, and then measured at the record head to produce the solid line in figure 1. The theoretical response of the geophone was calculated from the manufacturer's specifications and combined with the calculated response of the electronics. The combined response is minus 3db in amplitude at 4.5 hertz, but is essentially flat from 8 to 50 hertz. The square root circuitry was calibrated in a cold room to verify that it follows the proper power law. This data (figure 2) produced the regression relationship VOLTSout=SQRT(K*VOLTSin) for an input voltage of 10 millivolts to 10 volts. The constant factor, K, is 4.5.

A 5 hertz calibration signal was recorded on the OBS, digitized, and squared to show the type of waveform distortion generated by the analog playback head (figure 3). In addition, the digitized output of a perfect 2 hertz square wave (figure 4) demonstrates that the lowest harmonic has been phase shifted but the higher harmonics which combine to preserve the sharp onset, have not.

FILTER DESIGN

A filter can be realized by comparison of a distorted output with the original undistorted signal, in frequency space, simply by the division of the latter by the former. The frequency domain filter can then be inverted to a time domain filter, and applied to the signal by convolution to avoid the windowing problems of finite discrete transforms. When constructing this type of filter, it is helpful to have the discrete frequencies of the Fourier transform matched to the harmonics present in the test signal. This criterion avoids the need for vectorial addition of the Fourier frequencies near the test wave frequencies, and is satisfied by having a whole number of test wave cycles in the transform time window.

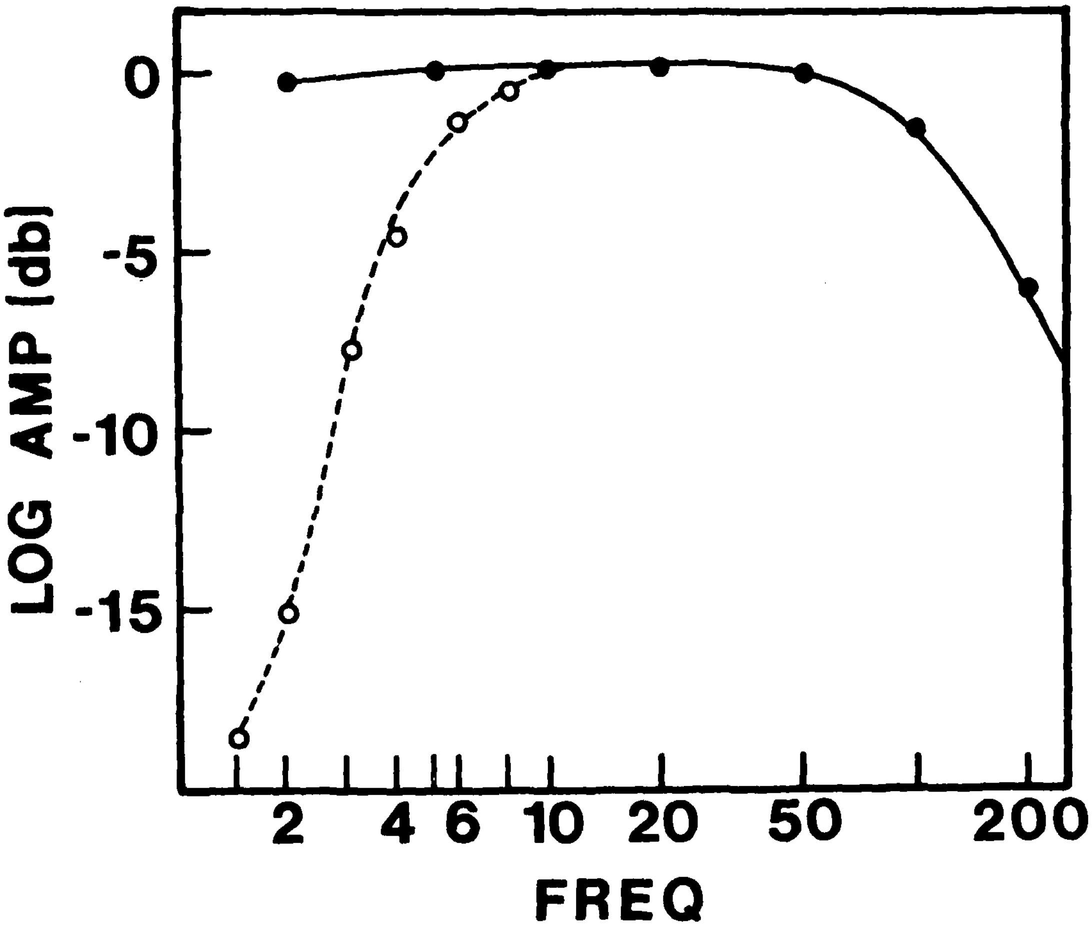

FIGURE 1. Measured frequency response of the OBS. The solid line is the response of the electronics. The dashed line is the calculated response of the electronics including the geophone sensor. Redrawn from L. Bond, unpublished technical report 1979.

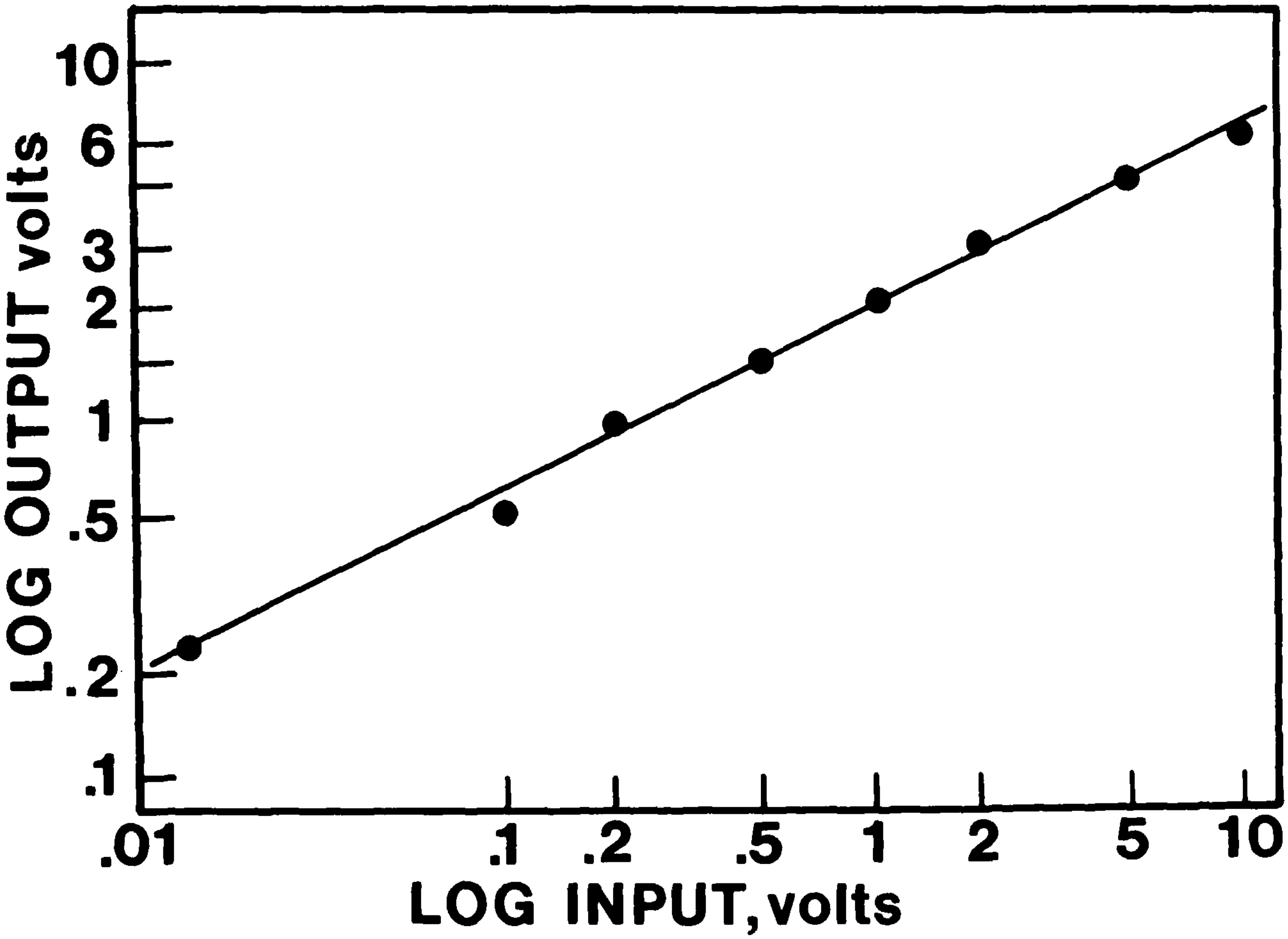

FIGURE 2. Measured transfer function of the OBS electronics including square-root circuitry. The data was taken at 5 hertz. Redrawn from L. Bond, unpublished technical report 1979.

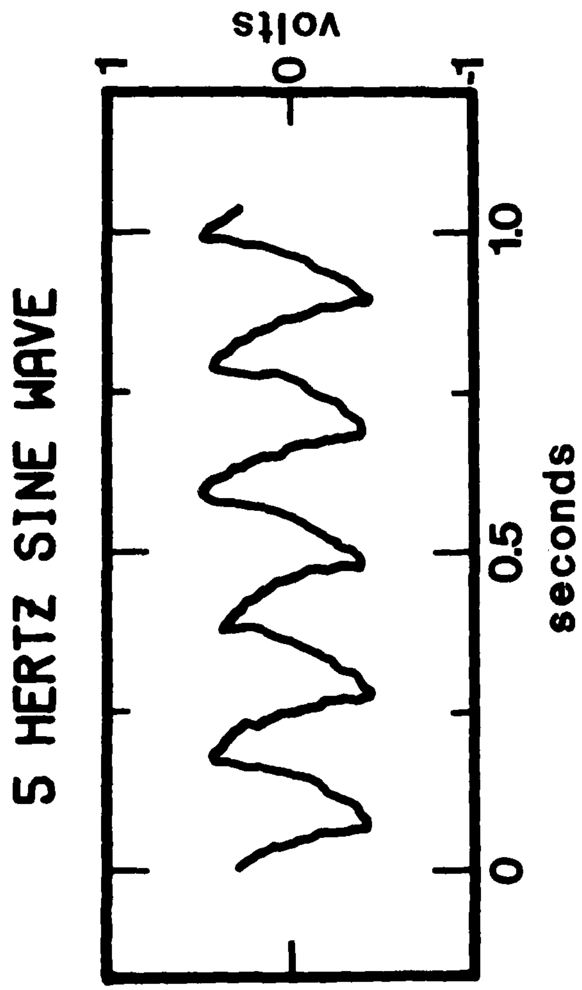

FIGURE 3. The digitized output of a 5 hertz calibration sine wave. Note scalloped waveform.

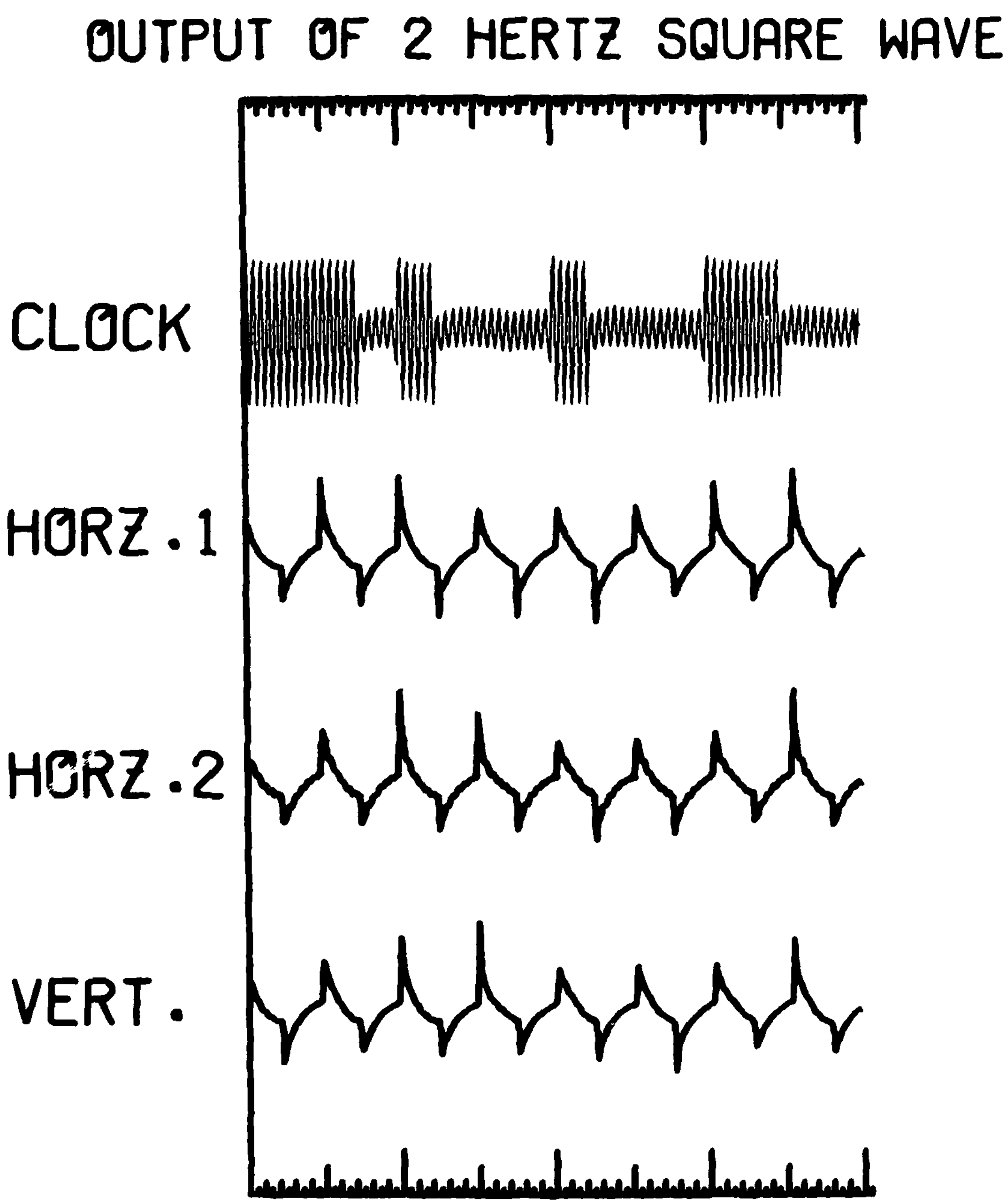

FIGURE 4. The digitized output of an inputted square wave. The identical signal was fed into all 3 channels.

PROPERTIES OF FOURIER TRANSFORM PAIRS

The Fourier pairs $f(t)$ amd $F(\omega)$ have properties that simplify computation in the frequency domain. A physically realizable time function, i.e. any geophysical record, is a pure real function, and the corresponding $F(\omega)$ is complex and is defined for both positive and negative ω up to the Nyquist frequency. Thus it contains redundant information, described by

$$F(\pm\omega) = F^*(\mp\omega)$$

This implies that the real part of $F(\omega)$ is even (the same for positive and negative ω) and the imaginary part is odd. A function fulfilling these conditions is called Hermitian. All geophysical spectra are Hermitian. Furthermore, since

$$F(\omega) = a(\omega) + i\,b(\omega) = |F(\omega)|e^{i\phi(\omega)}$$

then the modulus $F(\omega)$ is even and the phase $\phi(\omega)$ is odd.

The time-shifting theorem is also a useful property. Given that $f(t)$ transforms to $F(\omega)$, then for a real constant alpha (α),

$$f(t \pm \alpha) \longrightarrow e^{\pm i\alpha\omega} F(\omega) = |F(\omega)|e^{i(\phi(\omega) \pm \alpha\omega)}.$$

If a time function is shifted by a constant amount, then its amplitude spectrum remains unchanged while its phase spectrum changes by a linear amount determined by the time shift. This property is useful to control the position of the final time domain filter in the transform window.

EXPERIMENTAL PROCEDURE

An experiment was designed that would test the phase response of the playback electronics within the seismic bandwidth 2 to 25 hertz. The test signal was the clock's internal 2 kilohertz oscillator that was divided down by 1024 to produce an accurate, stable square wave of frequency 1.9531250 hertz. The number of sample points used in the Fast Fourier Transform (FFT) routine was 1024. This assured a whole number of cycles, 20, in the sample window. The frequencies present in a square wave are the fundamental plus an exponentially decreasing series of the odd harmonics. Thus, the frequencies sampled using this square wave are approximately 2,6,10...hertz. The test signal was inserted at the record head amplifier of all three channels but by-passed the other active electronics of the OBS. The clock signal was recorded to synchronize the digitization on playback.

COMPUTATION OF THE FILTER

The filter is constructed in the frequency domain utilizing only the fundamental and odd harmonic frequencies. The algebra of the filter is as follows. One can define the principal time functions as;

s(t)=input square wave
d(t)=distorted output
f(t)=inverse filter
g(t)=forward filter,

and in frequency space each time function Fourier transforms to,

s(t) becomes S(w)
d(t) becomes D(w)
f(t) becomes F(w).

Simple filter theory states that an input is convolved (*) with a filter to produce an output (equation 1).

$$s(t)*g(t)=d(t) \qquad (1)$$

An inverse filter, f(t), can be found such that

$$g(t)*f(t)=delta(t),$$

the unit spike function. Convolving f(t) with equation 1 produces

$$s(t)*g(t)*f(t)=d(t)*f(t).$$

Since a function convolved with a delta function is the original function, then

$$s(t)=d(t)*f(t). \qquad (2)$$

If a suitable filter can be constructed, then the original signal is recovered by convolving f(t) with the distorted output.

Convolution in one domain is equal to multiplication in the transform domain. Thus equation 2 can be directly written in Fourier space as,

$$\begin{aligned} S(w)&=D(w).F(w) \\ &\text{or} \\ F(w)&=S(w)/D(w). \end{aligned} \qquad (3)$$

The functions S(w), D(w), and F(w) are complex (equation 4) so that the filter F(w) can be expressed as a complex division (equation 5).

$$\begin{aligned} S(\omega)&=X_S+iY_S \\ D(\omega)&=X_D+iY_D \end{aligned} \qquad (4)$$

$$F(\omega)=\left[\frac{X_S\cdot X_D+Y_S\cdot Y_D}{X_D^2+Y_D^2}\right]+i\left[\frac{Y_S\cdot X_D-Y_D\cdot X_S}{X_D^2+Y_D^2}\right] \qquad (5)$$

The frequencies present in the test square wave are the odd harmonics of 1.953251 hertz. These frequencies are identical to the discrete Fourier frequencies calculated for 1024 points by the FFT program. Thus only the complex values at the harmonics are used in equation 5 to calculate the complex values of the filter.

The filter is also Hermitian and must satisfy the oddness/eveness properties in a smooth fashion. Figure 5 displays two partial plots of the imaginary values of the filter, near the edges of the frequency window where the amplitude of the imaginary part must go through zero. The circles are the discrete frequency values and the connecting line was calculated using a cubic spline interpolation routine. However, two alterations were made in generating the continuous version. First, a Gaussian curve was applied from 0 to 6 hertz with the 1.95 hertz value eliminated. This produced a smooth transition to zero hertz and prevented the filter from trying to compensate for the natural low frequency rolloff of the playback head. At the folding point or Nyquist frequency, the oddness requirement was satisfied by eliminating the 48.8 hertz value and by a similar Gaussian smoothing from 45 to 50 hertz. The real part of the filter was cubic spline interpolated between harmonic values with a smooth continuation to the folding point (figure 6).

The functions S(w) and D(w) can be alternatively expressed as Fourier series (equation 6).

$$\begin{aligned} S(\omega) &\sim S_\omega = \sum_n S_n e^{i(\omega_n + \theta_n^S)} \\ D(\omega) &\sim D_\omega = \sum_n D_n e^{i(\omega_n + \theta_n^D)} \end{aligned} \qquad (6)$$

The expression for the filter in equation 3 becomes,

$$F(\omega) \sim F_\omega = \sum_n \left(\frac{S_n}{D_n}\right) e^{i(\theta_n^S - \theta_n^D)}$$

The response of the filter is the ratio of the amplitude responses, Sn and Dn, and the difference in the phase responses. The amplitudes at the harmonics are practically equal (figure 7), and the ratio (Sn/Dn) is approximately unity. In other words, the filter does not alter the Fourier amplitudes in the signal significantly.

The phase response of a perfect square wave is a constant. However, an initial time shift will result in a linear slope in the phases. This slope is seen in the phase responses of Sw and Dw (figure 8). The difference in these phases (lower plot) is the response of the filter but with a different slope. The deviation from this slope is the phase angle correction that is applied to the signal. The phase distortion occurs in the seismic bandwidth 2 to 15 hertz (figure 8) and is most severe below 10 hertz.

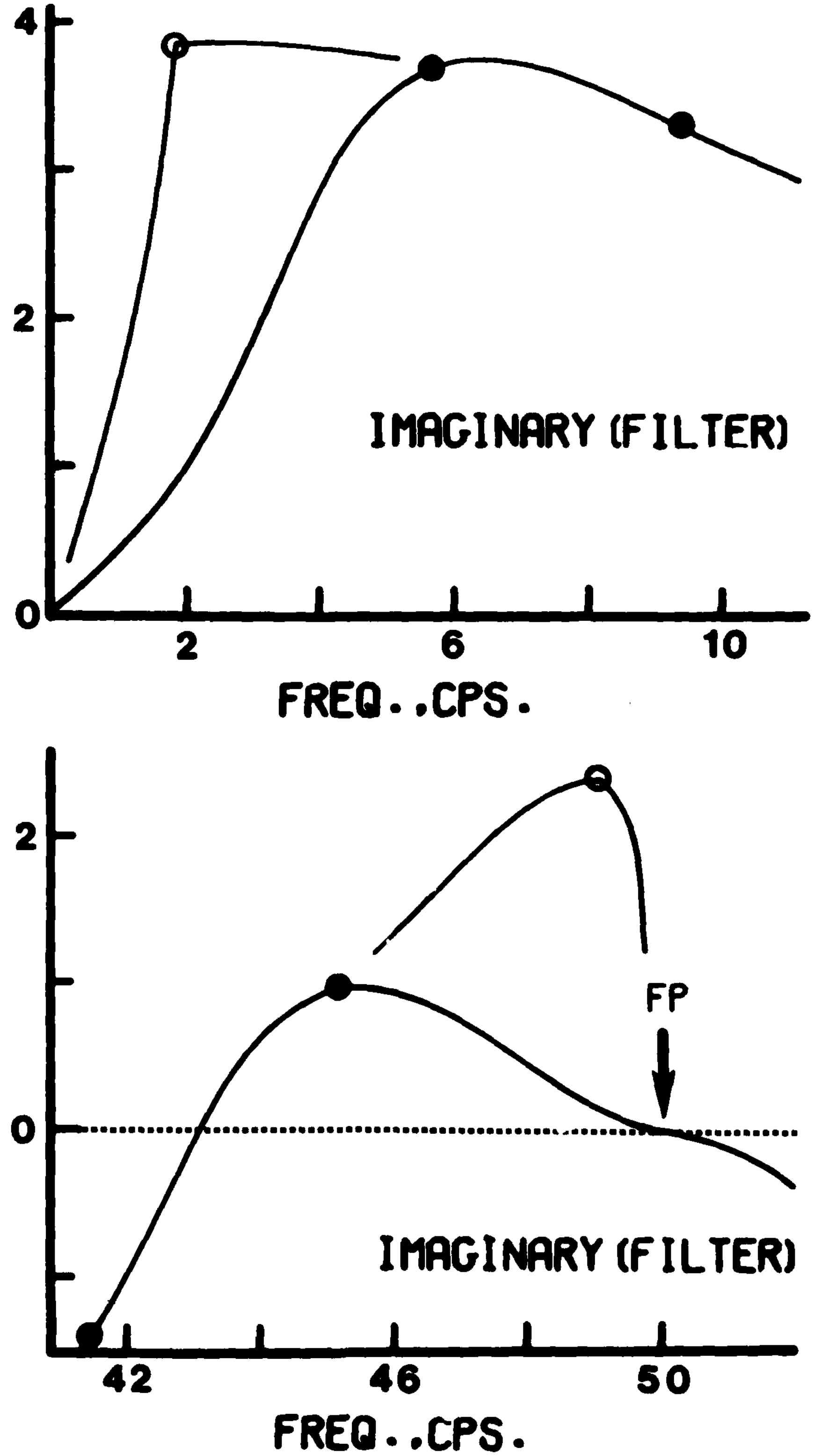

FIGURE 5. Plots of the imaginary part of the complex inverse filter. Ordinate units are arbitrary. See text for discussion.

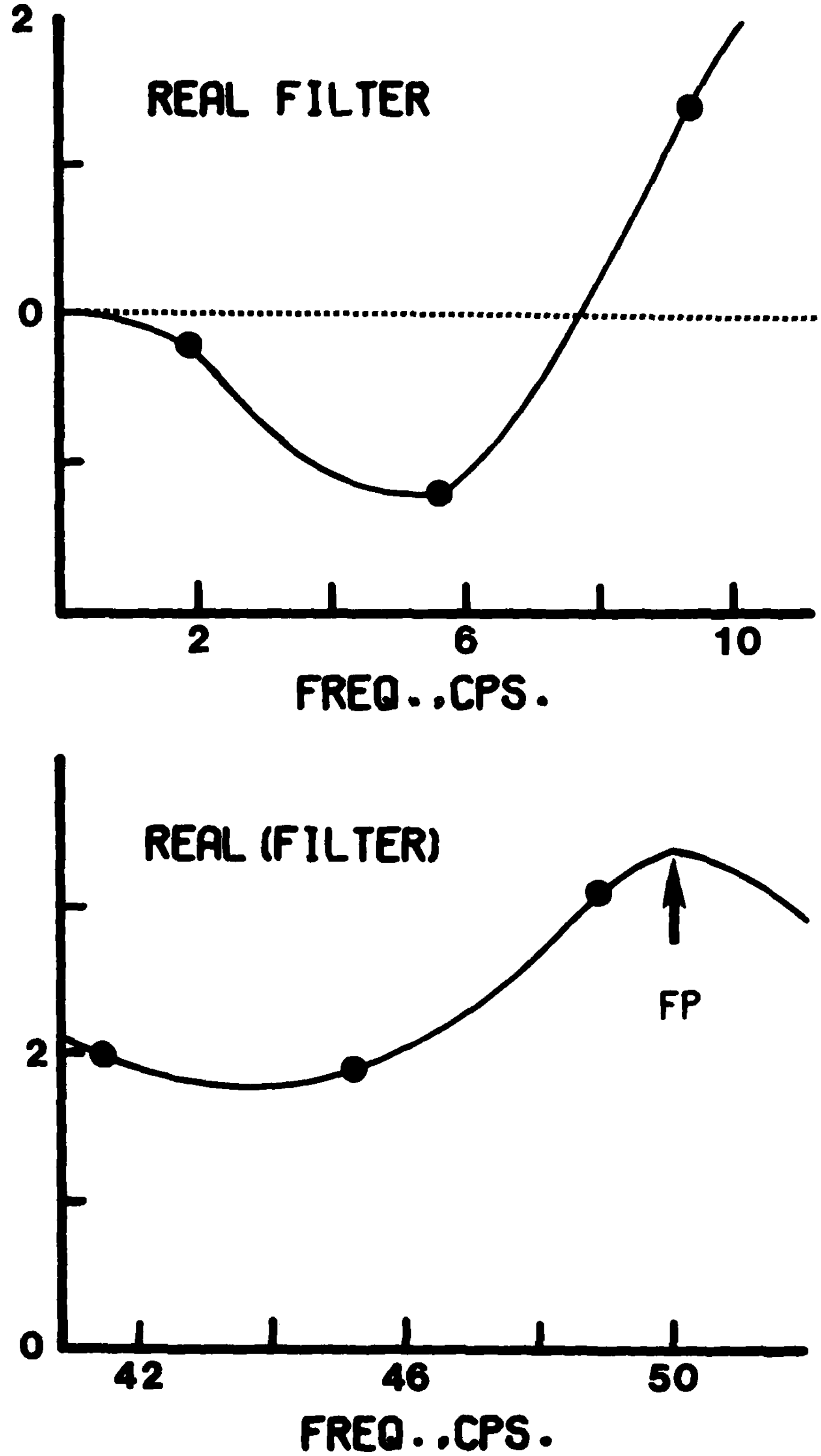

FIGURE 6 Plots of the real part of the complex inverse filter, near the edges of the frequency window.

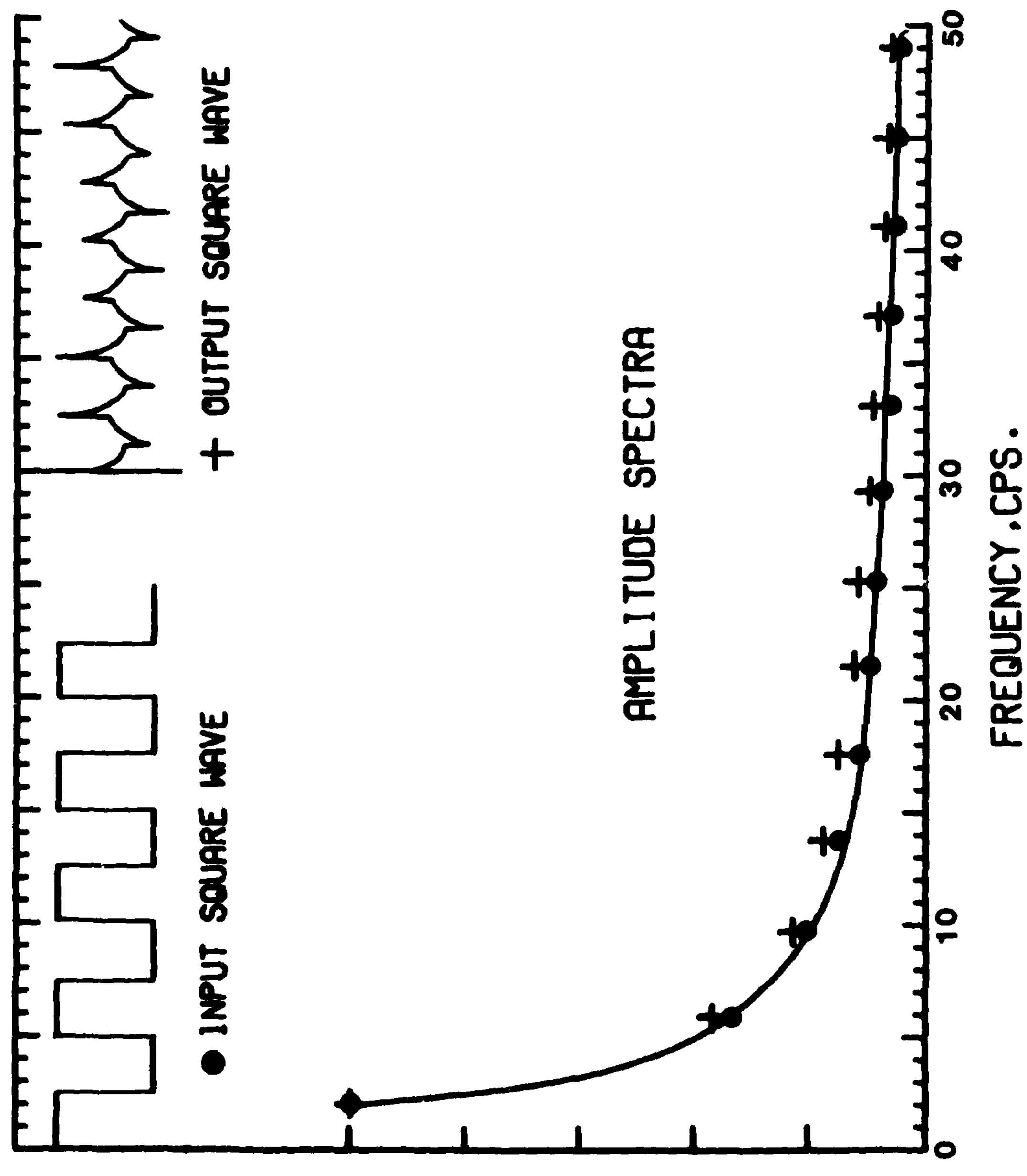

FIGURE 7. A comparison of the amplitude spectra of a perfect square wave (filled circles) with the spectra of the distorted output (crosses). The amplitude scale is linear and the plots have been normalized to 1.

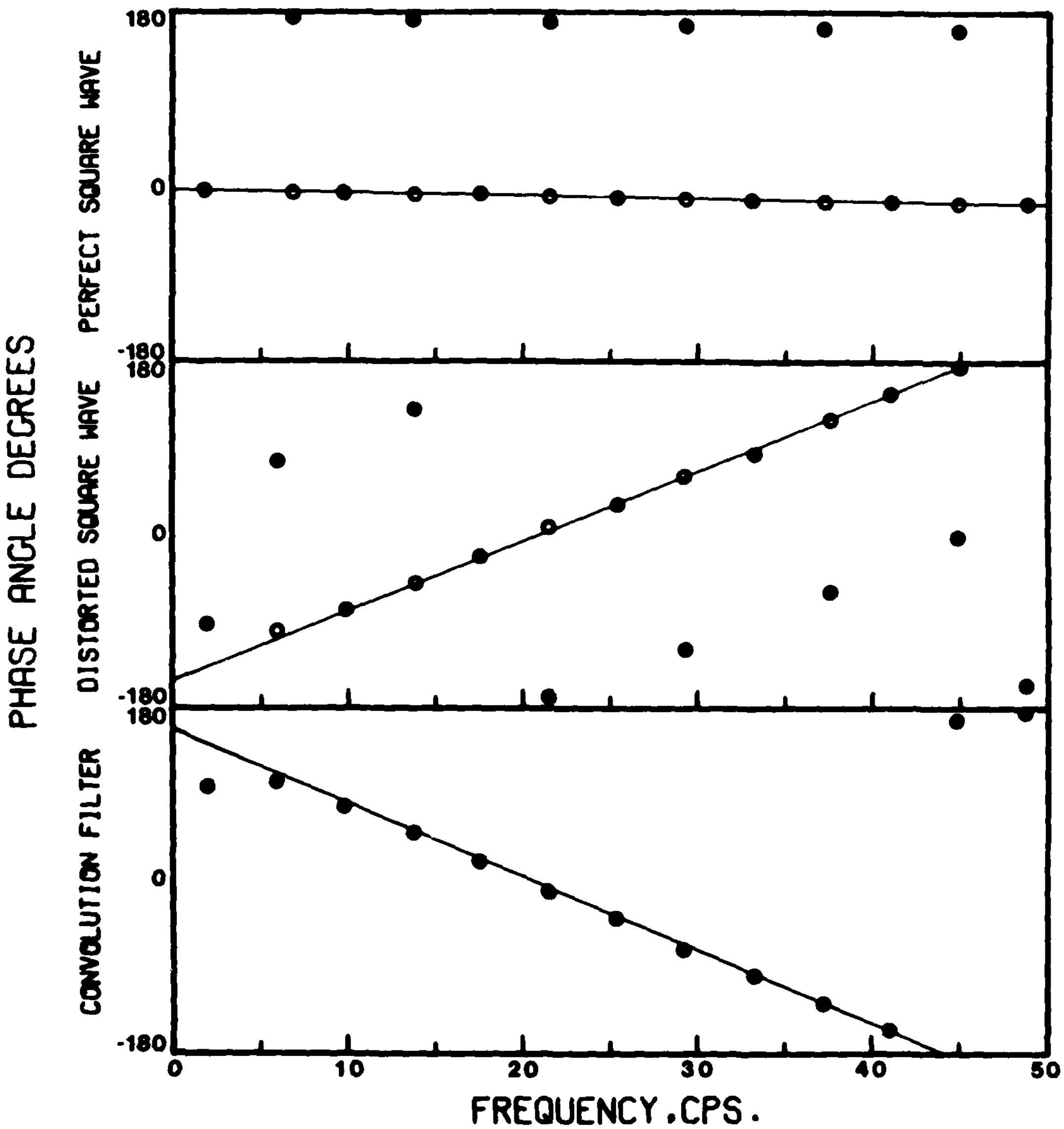

FIGURE 8. A comparison of the phase responses of a perfect square wave, the distorted output signal, and the final inverse phase filter over the bandwidth 0 to 50 hertz. EXPLAIN DOTS AND CIRCLES

TIME DOMAIN REPRESENTATION OF THE FILTER

The phase-correcting filter was constructed and modified in the frequency domain. It is inverted to a time domain function by an inverse Fourier transform and consists of 1024 points, the same number as in the frequency domain. Figure 9 is a plot of the middle 250 points of the filter. The energy is limited to the center points with the tails being near zero. The filter's amplitude spectrum can be regarded as flat, even on this exaggerated linear scale (figure 9). The center 50 points (0.5 seconds) of the filter retains greater than 99 percent of the total power of the original filter. This length filter can be effectively convolved with the signal to produce the desired results. In addition, the resultant 50 point operator is computationally efficient for moderate length data sets.

The final truncated filter is smoothed by a running 3-point binomial filter. This has the desired effect of low-pass filtering the inverse filter to remove any residual high frequency oscillations. However, the filter remains flat over the seismic bandwidth 5 to 20 hertz, and the response is minus 3db in amplitude at 3 and 22 hertz with a slow roll-off to the Nyquist frequency, 50 hertz.

RESULTS

A test of the filter was made by convolving the truncated version with the original distorted square wave. Figure 10 is a sketch of several cycles of the filtered square wave. The phase correction has been accomplished effectively. A very slight rounding and overshoot is present on the square wave edges due to the steps taken to smooth the filter, but the anomalies are smaller than the random variations due to tape and head-amplifier noise.

The final result obtained by convolving the phase correction filter with a seismic trace is displayed in figure 11 (trace PHASE). The initial trace, labeled RAW, is the digitized, square-rooted signal. The SQR-trace is the result of bi-polar squaring the RAW-signal. The waveform is overall spiky in nature and the first-energy arrival time is a half-cycle late, about 0.05 seconds. Additional traces

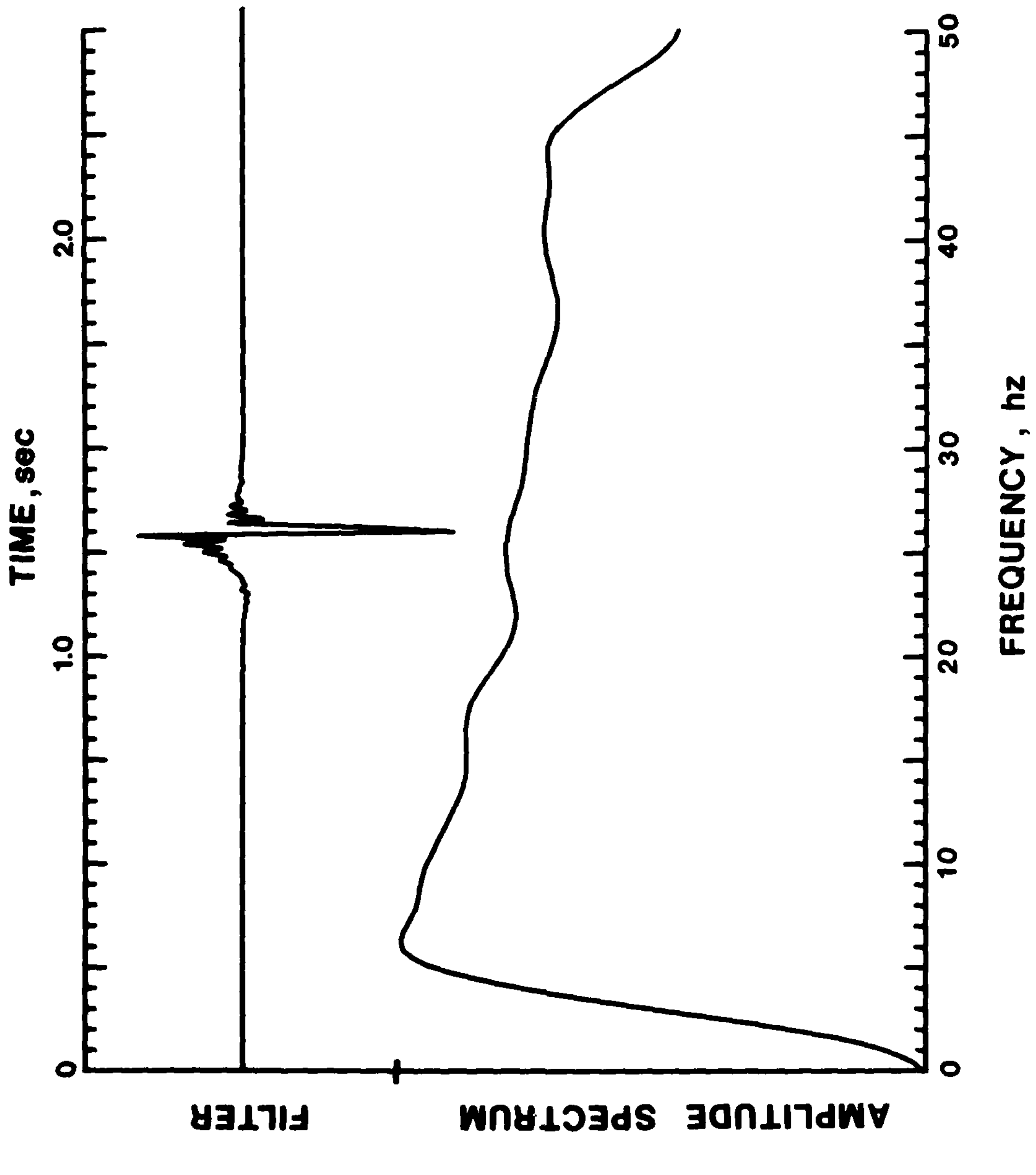

FIGURE 9. A linear plot of the amplitude spectra of the inverse phase filter

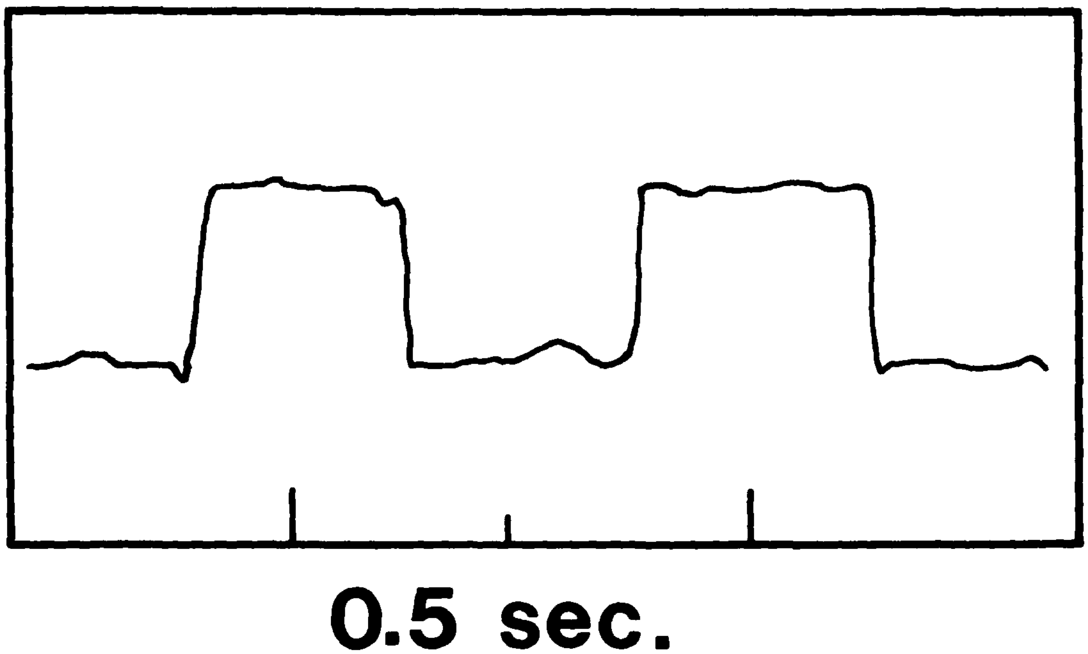

FIGURE 10. A sketch of the waveform that results from convolving the distorted square wave signal with the inverse phase filter.

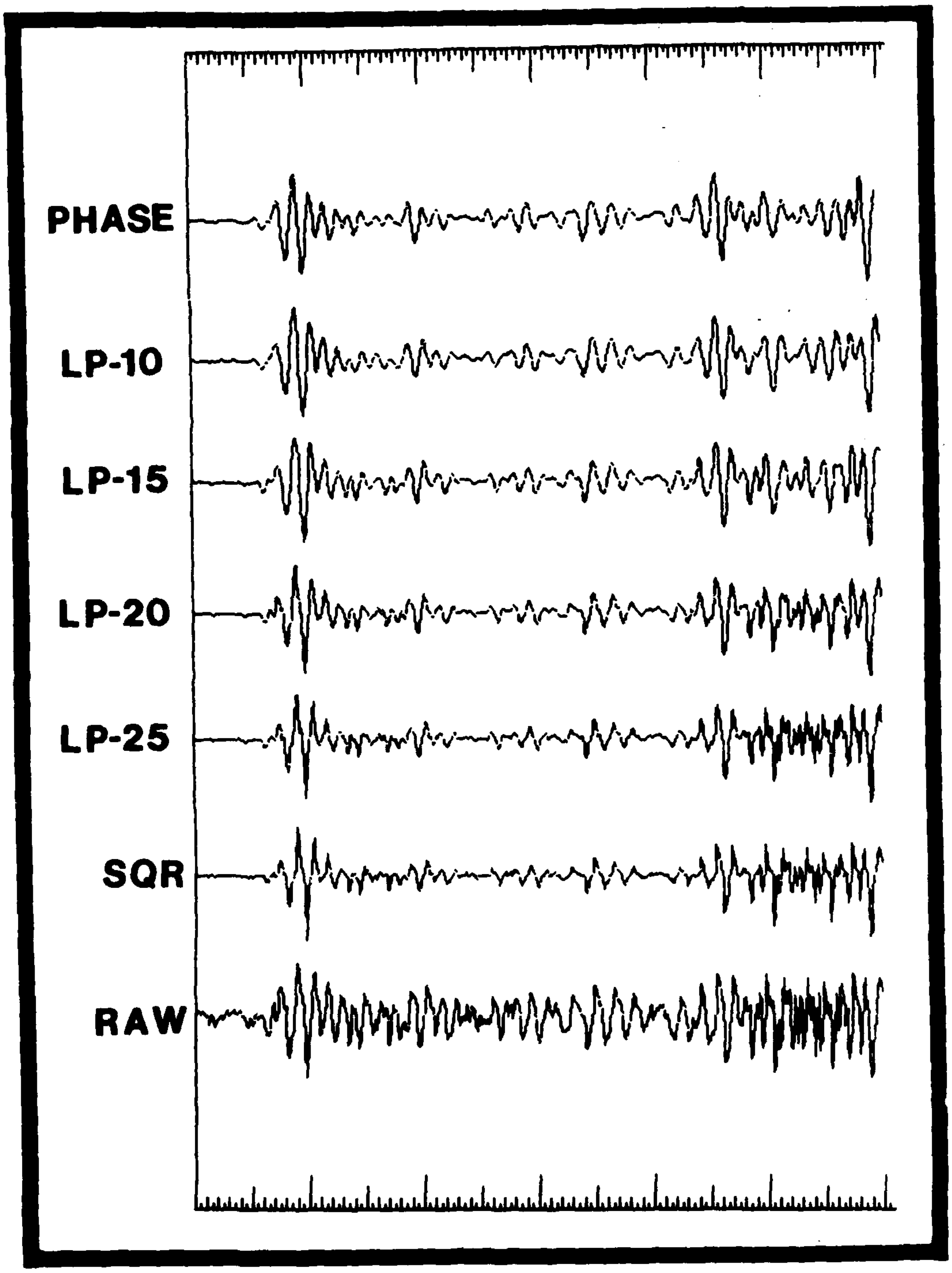

FIGURE 11. A comparison between the RAW seismic signal and the phase corrected signal, trace PHASE. The SQR-trace is the raw signal squared, and the LP-traces are various low-pass filtered signals. Refer to text for discussion.

were generated from the raw squared data by progressively filtering out more and more of the high frequencies, consisting mostly, but not exclusively, of spurious harmonics of the basic seismic signal. They are designated LP- and the -3db cutoff point in hertz of a zero-phase shift low-pass convolution filter less than 0.5 seconds in length. The difference between these LP-signals and trace-PHASE are subtle but significant. The general waveform progresses from being very spiky (LP-25) to being overly 'filtered' looking (LP-10). The first motion of the PHASE-trace is a distinct positive (upward) pulse. Only trace LP-10 begins to show the first arrival properly, but it is still several hundreths of a second late. It should be noted that the amplitudes for each trace are relative to a maximum excursion of one inch, and therefore can not be used for direct comparison. However, the amplitude of the first arrival wave packet of the LP-traces were usually no more than 80 percent of that of the PHASE-trace.

CONCLUSIONS

Until recently nearly all the results obtained from marine seismic refraction have been based on the analysis of the travel-time distance relations for the most prominent arrivals on the records. In this report, and its companion (Determination of the orientation and gain of horizontal geophones used in ocean bottom seismometers, Technical Report Number 386) new methods have been described which attempt to obtain a more detailed model of the velocity structure by making use of more of the information on the seismograms. The combinations of travel-time analysis with studies of the true amplitudes, particle motions and waveforms of the arrivals, places much greater constraints on the velocity structure and increases the potential resolution of the refraction technique. In the future, the spectral content of an arrival may be as valuable in deducing detailed structure as the timing of that arrival, or its raw amplitude. To obtain this information, it is essential that the characteristics of the receiver and playback electronics be well known and corrected for in processing as well as practicable.

The unprocessed signal from the University of Washington OBS showed that some form of distortion was occuring in the seismic bandwidth. By recording a synchronised square wave, and comparing the output with the input in the frequency domain, it became obvious that the lower harmonics of the square wave were being phase shifted. A phase correcting filter was constructed in the frequency domain and inverted to a time domain convolution operator of 50 points (0.5 second) length.

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