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BLACS — A NEW CORRECTION SCHEME OF ANALOG ACCELEROGRAMS. PART – 1: DETAILS OF SCHEME

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Abstract

In this paper a correction scheme developed in frequency domain and named as Band Limited Accelerogram Correction Scheme *(BLACS)*, is presented for processing the records of analog accelerograph . The scheme has complete control on the frequency contents of the digitised sequence and first performs band limited interpolation of the nonuniformly spaced samples to get the sequence at 200 *SPS*. It has the provision to decimate the data to 100 *SPS* taking account of band limited property. The instrument correction and band pass filtering is then performed in frequency domain. The band pass filtering is done by using $|H(j\omega)|^2$ of a Butterworth filter which does not introduce any phase shift. In case the corrected accelerogram is required at a higher sampling rate then the scheme performs band limited interpolation to achieve the desired sampling rate. In the second part of this paper a comparison of this scheme is done with that of schemes of Lee and Trifunac, Erdik and Kubin and that of Khemici and Chiang.

INTRODUCTION

Based on the work published by Kumar *et. al* (1992) and Basu *et. al* (1992) on band limited interpolation of nonuniform and uniform samples respectivily, a new correction scheme named *BLACS* for processing records of analog accelerographs is presented. This scheme first converts the digitised data into time *vs.* acceleration values using the calibrations to get an uncorrected accelerogram. If the uncorrected accelerogram has nonuniform samples, then band limited interpolation is performed to recover the data at 200 *SPS*. The data is then decimated to 100 *SPS* in such a way as to preserve the frequency content upto the new Nyquist frequency (50 *Hz*). The instrument correction is applied in frequency domain by assuming the accelerometer to be a single degree of freeedom system. The signal is then band-pass filtered in frequency domain using transfer function of the Butterworth filter for obtaining the corrected accelerogram. The velocity and displacement histories are obtained by integrating the corrected accelerogram in frequency domain. In case the accelerogram is required at higher sampling rate, the band limited interpolation as described by Basu *et. al* (1992) is employed. Figure 1 gives flow chart of *BLACS* and various steps of the flow chart are discussed in detail.

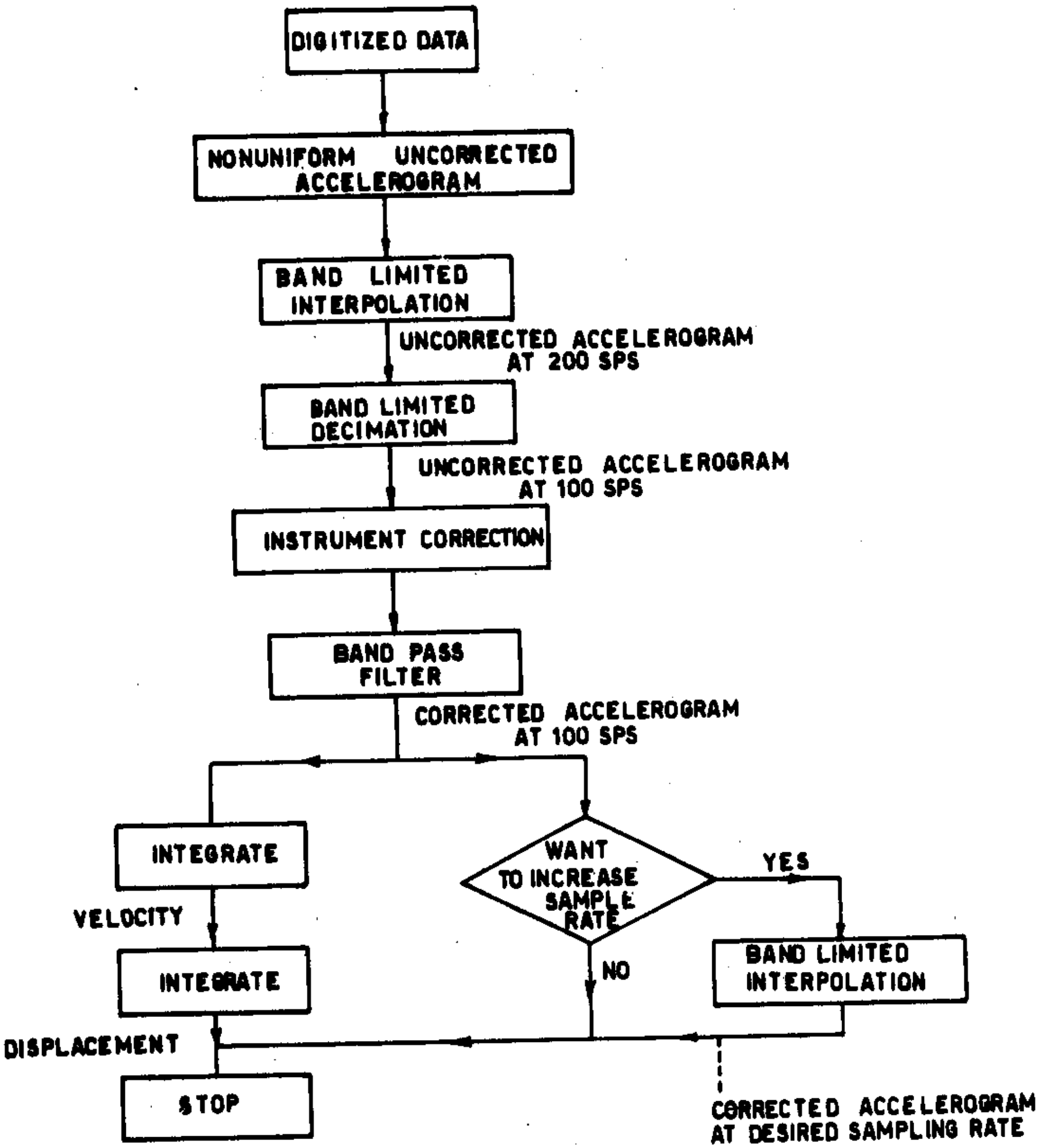

Figure 1: Flow chart of BLACS.

Some of these steps are examined by imparting white noise input and comparing its output with an ideal situation. It may be mentioned that *BLACS* comprises of steps as discussed above and some of the steps may not be required for a particular case and therefore can be dropped. A judgement in this regard will have to be made by the users. However, for the purpose of discussion in this paper, the scheme is assumed to be comprising of all the steps.

THE UNCORRECTED ACCELEROGRAM

An earthquake record of an analog accelerograph generally has six traces. Three of these are records of longitudinal, transverse and vertical components of ground acceleration while two fixed traces are from mirrors rigidly attached to the accelerograph's frame. These fixed traces are used to neutralise the low frequency noise caused in the accelerograms due to paper (or film) distortion, motion of the film in the drive mechanism etc. The sixth trace is of quarter second OFF and quarter second ON time mark spots which provide the time axis of the record. The time mark spots are essential as the photographic paper may not have a completely uniform and constant speed. Thus, when an earthquake record of an analog accelerograph is digitized on a digitizing table, the operator generates six different files — one each for the six traces. For processing of each of the components; the time mark file, one of the fixed trace file (generally one which is nearest to the component) and the file of the concerned component, are used. All these files provide abscissa and the ordinate of points digitized.

The first step in processing the digitized record is to fit a least square line through the fixed trace to determine the angle from the abscissa of digitizing table. This is required as the accelerogram and table axes may not be exactly parallel. A correction for the rotation of the coordinates is then applied to all the three files. Subsequently, the ordinates of earthquake file are subtracted from the corresponding ordinate at the same abscissa of the fixed trace file. The corresponding ordinate at the same abscissa of fixed trace file is found through linear interpolation. Next a least square line is fitted in the above processed earthquake file which provides the ordinate of the origin of the new coordinate system. The abscissa of this origin is the abscissa of the first point of the earthquake record. The developed earthquake file is then translated to the new coordinate system. This provides the uncorrected accelerogram at nonuniform sampling interval. Figure 2 gives flow chart for getting uncorrected accelerogram from the digitized data.

RECOVERY OF SIGNAL FROM NONUNIFORM SAMPLES

This step is used if the digitization is performed on semi-automatic digitizer from which the sequence is obtained at nonuniform interval. However, if the digitization is performed in an automatic scanner then this step is not required.

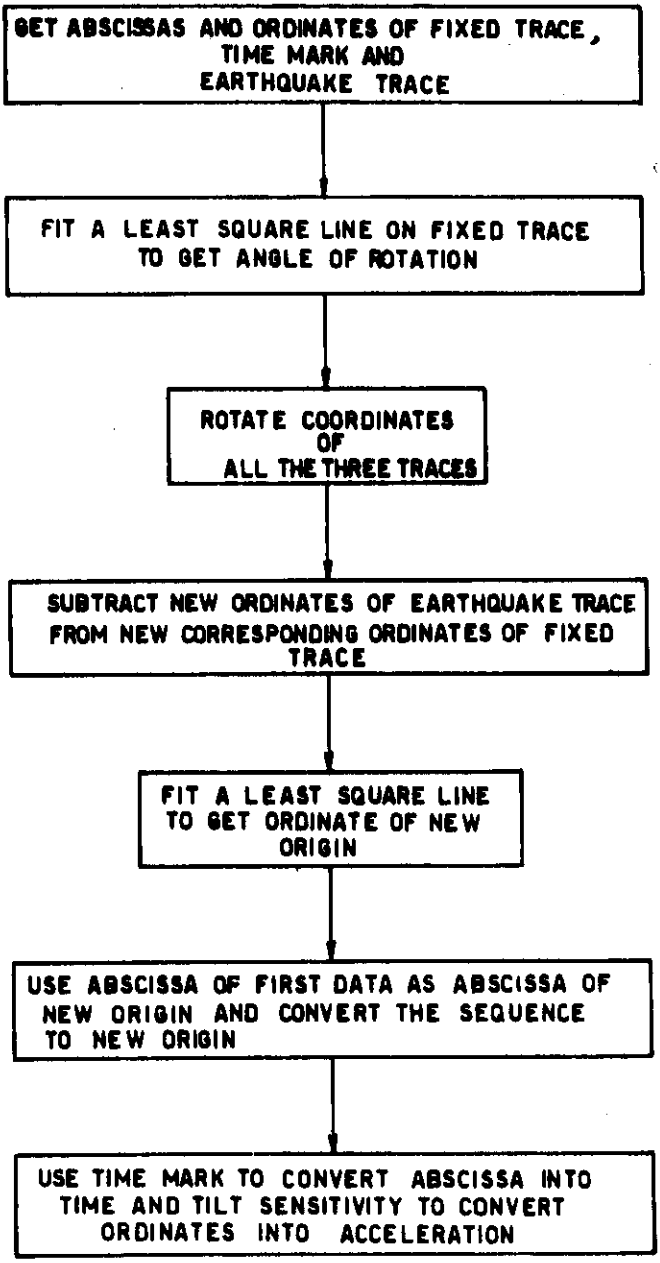

Figure 2: Flow chart for getting uncorrected accelerogram from the digitized data.

The signal is required to be recovered from given sequence at nonuniform interval in such a way so as to maintain the same frequency content. The word recovery of signal in this study means getting digital data at comparativily large sampling points at constant interval. In this work, the signal is recovered at 200 *SPS* .

Marvasti *et al.* (1991) have shown that for nonuniform samples given by

$$x_s(t) = \sum_i x(t_i)\delta(t - t_i) \quad (1)$$

where $\{t_i\}$ is a stable sampling set, the following iterative method shall recover the original band limited signal $x(t)$ from $x_s(t)$

$$x_{k+1}(t) = \lambda PS(x(t)) + (P - \lambda PS)x_k(t) \quad (2)$$

where λ is a convergence constant whose value should be between 0.5 and 1, $x(t)$ is the original signal and $x_k(t)$ is the signal obtained after k^{th} iteration. P is a band limiting operator (a low pass filter) and S is ideal nonuniform sampling operator or a staircase function operator. The band limiting operator P is self adjoint which means $P(x_k(t)) = x_k(t)$. The staircase function operator in this case assumes zero order hold of nonuniform samples and converts the nonuniform samples into a staircase form at a constant interval with 200 *SPS* . $PSx(t)$ in Equation 2 is low passed signal of staircase nonuniform samples which is known before the start of iterations. Marvasti *et al.* (1991) have also proved that there exist a range of values of λ for which the process of Equation 2 will converge i.e.

$$\lim_{k\to\infty} x_k(t) = x(t) \quad (3)$$

Figure 3 gives flow chart of the iteration used to recover the signal at 200 SPS from the given nonuniformly spaced data. Using above iterative scheme, uncorrected accelerogram at nonuniform sample interval is interpolated to 200 SPS. This process is discussed in detail by Kumar *et. al* (1992). The cutoff frequency used in the interpolation is half of average sampling rate of uncorrected accelerogram or 25 Hz whichever is lower. The interpolation is done using convergence constant $\lambda = 1$. Mean square error(*MSE*) determined between results of two consecutive iterations is used as convergence criterion and the iteration is assumed to have converged when *MSE* becomes less than 0.01.

DECIMATION

The band limited uncorrected data which is recovered at 200 SPS as described in the preceding section or which has been obtained directly at uniform sampling rate (in case of records digitized through automatic scanner) are decimated to 100 SPS data. The decimation is performed in a manner such that the frequency contents upto the new Nyquist frequency are preserved. To achieve this, FFT of the 200 SPS signal is taken (Nyquist frquency 100 Hz). The new Nyquist frequency is determined in proportion to the desired sampling rate. Thus in this case the new Nyquist frequency becomes

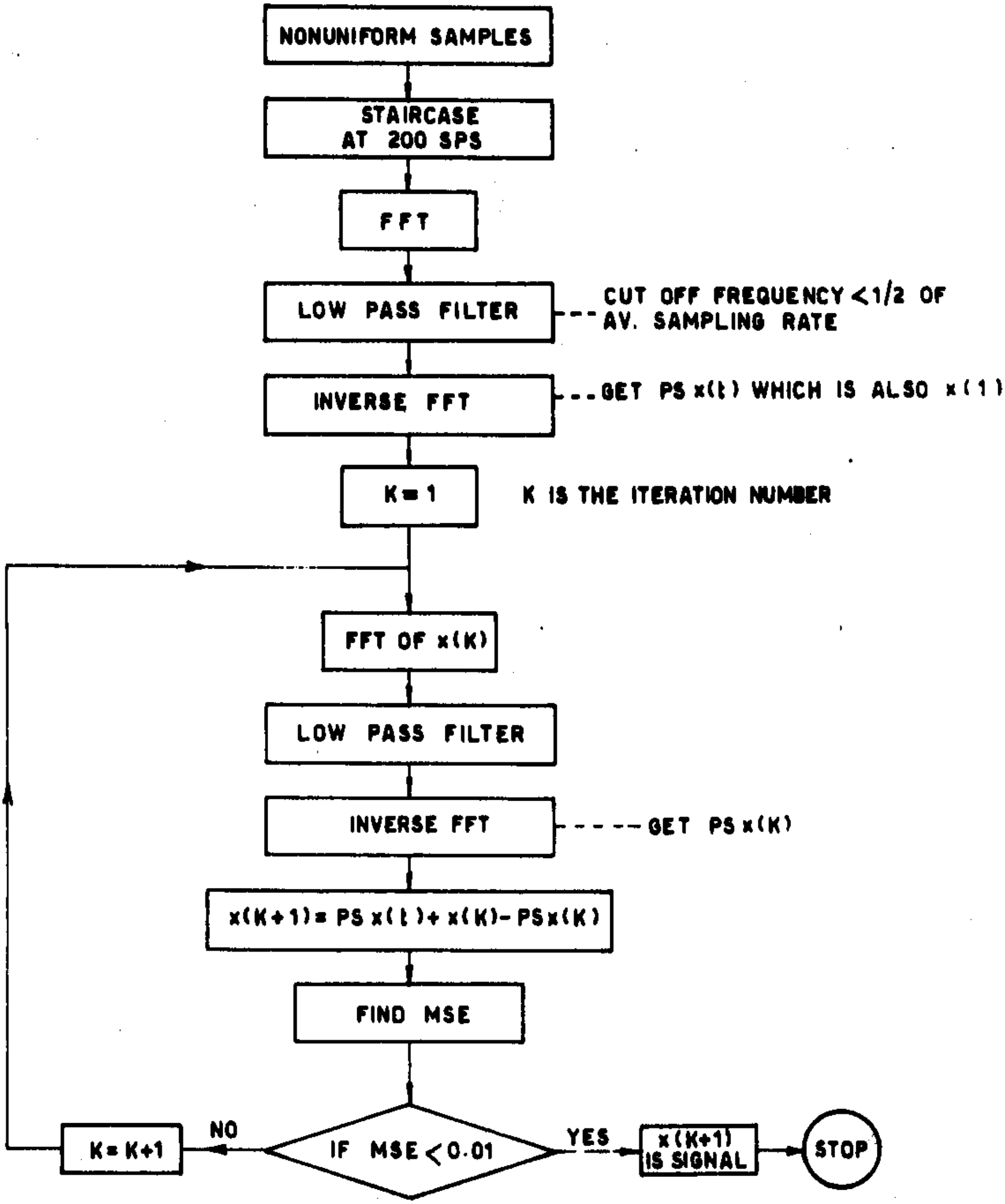

Figure 3: Flow chart of iteration for recovery of signal from nonuniform samples.

50 Hz. From the FFT of 200 SPS data, the real and imaginary parts from DC to new Nyquist frequency only are retained. The real parts from new Nyquist frequency to twice the new Nyquist frequency are taken as mirror image of real part from DC to new Nyquist frequency whereas the imaginary parts are taken as negative of mirror image of the corresponding imaginary part. A Fourier inversion of this set of real and imaginary data yields decimated history which preserves the frequency contents of the original signal upto the new Nyquist frequency. It may be noted that decimation reduces the total number of samples in proportion to the sampling rate. Thus, after decimation the bin frequency of the FFT remains the same but the total number of points in frequency domain as well as in time domain reduce in proportion to the sampling rate. In this work, the algorithm developed for the decimation works well when the sample reduction factor is a power of 2. The proposed band limited decimation should be performed with caution in case digitization of data is done on an automatic scanner, as it may introduce Gibb's phenomenon. In such cases data should be first low pass filtered with a cutoff frequency of maximum 50 Hz before performing the decimation. But, in case the 200 SPS data is obtained from band limited interpolation of nonuniform samples, the proposed band limited decimation would work well since such data is already low pass filtered.

INSTRUMENT CORRECTION

An uncorrected accelerogram may not require correction for instrument if the natural frequency of pedulum of accelerometer is substantially higher (about twice) than the largest frequency content of earthquake (generally assumed as 25 Hz) and the damping is fixed at 70% of critical (to get linear phase shift with respect to frequencies). However, the natural frequency of accelerometers used in the analog accelerographs cannot have a such high value as it will drastically reduce the sensitivity of instrument. The natural frequency of accelerometers of analog accelerographs usually varies from 18 Hz to 25 Hz and damping values in the accelerometers generally vary between 50% to 70% of critical. With this background, it becomes essential to perform instrument correction so that the recorded motion can be deconvoluted, to obtain ground excitation. This deconvolution in the present scheme is performed in the frequency domain.

If ω_n is the natural frequency in *radians/sec*, ς the fraction of critical damping of accelerometer, $x(t)$ the relative displacement of the pendulum and $a(t)$ the ground acceleration. Then the equation of motion of the pendulum of accelerometer is written as

$$\ddot{x}(t) + 2\omega_n\varsigma\dot{x}(t) + \omega_n^2 x(t) = -a(t) \tag{4}$$

where $\dot{x}$ and $\ddot{x}$ are the first and second derivatives of relative displacement of the pendulum. It may be mentioned that the values of acceleration in the uncorrected accelerograms are really $-\omega_n^2 x(t)$ which has been assumed to be equal to the ground acceleration $a(t)$. In other words, while deriving uncorrected accelerogram, the effect of first two terms of Eq. 4 are neglected. To get the frequency domain description from the given time domain sequence, let

$$x(t) = X(\omega)e^{j\omega t} \tag{5}$$

and

$$a(t) = A(\omega)e^{j\omega t} \tag{6}$$

where ω is the frequency contents of the signal in *radians/second* and $X(\omega)$ and $A(\omega)$ are the complex functions comprising real and imaginary parts of the Fourier transforms of $x(t)$ and $a(t)$ respectivily. By using Eqs. 5 and 6 into Eq. 4 one obtains

$$-\omega^2 X(\omega) + 2j\omega_n\varsigma\omega X(\omega) + \omega_n^2 X(\omega) = -A(\omega) \tag{7}$$

Equation 7 can be rewritten in the form

$$A(\omega) = -\omega_n^2 X(\omega)[1 - (\frac{\omega}{\omega_n})^2 + 2j\varsigma\frac{\omega}{\omega_n}] \tag{8}$$

in which $-\omega_n^2 X(\omega)$ on the right hand side of Eq. 8 is the Fourier transform of uncorrected accelerogram and calling

$$u(\omega) = \Re\{-\omega_n^2 X(\omega)\} \tag{9}$$

and

$$v(\omega) = \Im\{-\omega_n^2 X(\omega)\} \tag{10}$$

where $\Re$ is real part and $\Im$ is the imaginary parts of the discrete Fourier transform of $-\omega_n^2 x(t)$. Also in Eq. 8, the term inside the large bracket is the transfer function of the process of instrument correction $H(\omega)$ and let

$$p(\omega) = \Re\{H(\omega)\} = 1 - (\frac{\omega}{\omega_n})^2 \tag{11}$$

and

$$q(\omega) = \Im\{H(\omega)\} = 2\varsigma\frac{\omega}{\omega_n} \tag{12}$$

so that $A(\omega)$ can be written as

$$A(\omega) = [u(\omega) + jv(\omega)][p(\omega) + jq(\omega)] \tag{13}$$

With this Eq. 13 yields

$$\Re\{A(\omega)\} = u(\omega)p(\omega) - v(\omega)q(\omega) \tag{14}$$

and

$$\Im\{A(\omega)\} = u(\omega)q(\omega) + v(\omega)p(\omega) \tag{15}$$

An inverse Fourier transformation of the real and imginary parts of $A(\omega)$ gives the instrument corrected data from uncorrected accelerogram. For the sake of clarity this process of instrument correction is summarized as follows:

1. Find FFT of uncorrected accelerogram (which is now available at 100 SPS) to get $u(\omega)$ and $v(\omega)$. The bin frequency interval at which the real and imaginary parts of the FFT are determined will be equal to $100/N$ where N is the total number of sample points which is made equal to 2^k (where k is an integer) by inserting required number of zeros at the end of the sequence.

2. At the above bin frequency interval, determine the real and imaginary parts of the transfer function $H(\omega)$ *ie.* $p(\omega)$ and $q(\omega)$. Determine these values upto the Nyquist frequency (upto bin number $\frac{N}{2}+1$). For bin numbers $\frac{N}{2}+2$ to N, take $p(\omega)$ as mirror image of $p(\omega)$ determined upto Nyquist frequency. For bin numbers $\frac{N}{2}+2$ to N, take $q(\omega)$ as negative of mirror image of $q(\omega)$ determined upto Nyquist frequency.

3. Find real and imaginary parts of $A(\omega)$ from Eqs. 14 and 15.

4. Take inverse FFT to obtain time history of ground acceleration.

Response to White Noise

An exact white noise whose Fourier magnitude plot is a horisontal straight line with a magnitude of unity is generated at a sampling rate of 100 SPS and has 2048 data points *i.e.* it has a duration of 20.48 seconds. With FFT algorithm and availabilty of generation of random numbers in computer, generation of an exact white noise has considerably simplified. For a white noise of 2048 data points, any random number between -1 and 1 is picked up as the real part. Its corresponding imaginary part is found such that the magnitude is 1. Similarily 1025 pairs of real and imaginary parts corresponding to first 1025 bin frequencies (upto Nyquist frequency) are found. For bin numbers 1026 to 2048, real parts are found by taking mirror image of real part of the set upto Nyquist frequency and imaginary parts are found by taking negative of mirror image of imaginary parts of the set upto Nyquist frequency. Inverse FFT is then performed on these 2048 pairs of real and imaginary parts which yields the desired white noise. The signal so generated is also called *syncronous noise* and is used in digital signal processing to test filters and other processes (Harris,1987).

This white noise is used as input for the instrument correction. The natural frequency of the accelerometer is taken as 20 Hz and damping is taken as 60% of critical. FFT of the instrument corrected history is then determined. Figure 4 shows the Fourier magnitude plot upto 30 Hz of the instrument corrected history. This Fourier magnitude plot matches exactly with the ideal plot in the entire range upto the Nyquist frequency. This test shows the superiority of performing the instrument correction in the frequency domain.

BAND PASS FILTER

The next operation in the correction scheme is to perform the band pass filtering to remove low frequency and high frequency noise from the available sequence. Most of the high frequency noise has already been removed while recovering the signal during band limited interpolation of nonuniform samples (or at the time of decimation in case of data obtained from automatic scanners). However, to remove some of the high frequency noise which might have amplified during instrument correction, the low pass

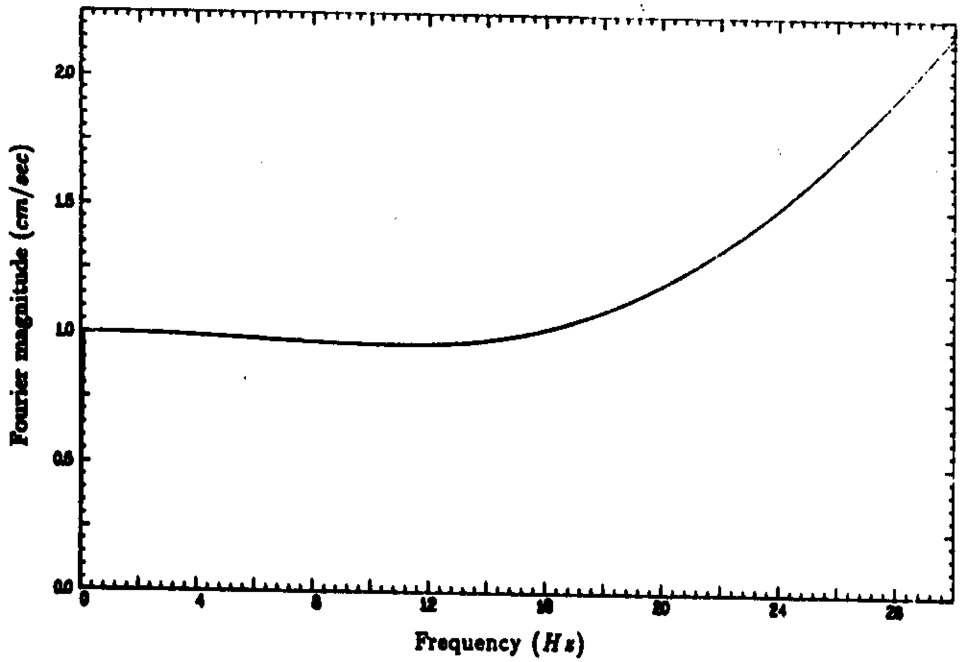

Figure 4: Response of proposed instrument correction scheme to white noise input.

operation is performed again alongwith high pass operation to remove low frequency noise by designing a band pass filter function corresponding to $|H(j\omega)|^2$ of Butterworth filter (Lam, 1979).

The filtering operation is performed by first finding FFT of the signal which is to be filtered. The bin frequency interval (Δf) at which the FFT of the signal is determined is given by

$$\Delta f = \frac{SPS}{N} \tag{16}$$

where SPS is the sampling rate which is 100 SPS in this case and N is number of data points of the sequence which is lengthened by inserting zeros so that the sequence has exactly 2^j (where j is an integer) data points. For example if sequence has 3910 data points then N is made 4096 (which is 2^{12}) by inserting 186 zeros in the sequence. At the above bin frequency intervals and upto the Nyquist frequency, transfer function $|H_1(j\omega)|^2$ of Butterworth filter for low pass operation, which is given by the equation below, is determined.

$$|H_1(j\omega)|^2 = \frac{1}{1 - (\frac{\omega}{\omega_l})^{2n}} \tag{17}$$

where ω is the frequency at which the function is determined, ω_l is the cutoff frequency of low pass filter and n is the order of filter. It is clear that cutoff frequency of the lowpass filter should be less than half of average sampling rate of the original nonuniform sampled data or 25 Hz whichever is smaller.

Similarily for high pass operation, transfer function $|H_2(j\omega)|^2$ of Butterworth filter is determined at all the bin frequencies upto Nyquist frequency. However, now the transfer function will be given by

$$|H_2(j\omega)|^2 = \frac{1}{1-(\frac{\omega_h}{\omega})^{2n}} \tag{18}$$

where ω_h is the cut off frequency of the high pass filter.

The net transfer function of the band pass filter $|H(j\omega)|^2$ at each bin frequency is determined by

$$|H(j\omega)|^2 = |H_1(j\omega)|^2\,|H_2(j\omega)|^2 \tag{19}$$

Transfer function of the band pass filter from Nyquist frequency to twice of Nyquist frequency is found by taking the mirror image of the transfer function from DC to Nyquist frequency.

Convolution is next performed between the FFT of the signal and the transfer function of the filter. As the transfer function of filter has only real part, convolution in frequency domain means simple multiplication of real and imaginary parts of the Fourier transform of the signal with the filter function at the corresponding bin frequencies. Inverse FFT is then performed to get the band passed signal in time domain which gives the corrected accelerogram at 100 SPS.

Again to check the performance of this band pass filter, its response for white noise input of 100 SPS is determined. The cutoff frequency for the low pass filter is taken as 25 Hz and for high pass filter is taken as 0.1 Hz. The Fourier magnitude of the band pass filtered white noise is plotted and is shown in Fig. 5 . In this figure, the tapering of high pass operation is not properly seen. This is plotted by zooming Fig. 5 and is shown in Fig. 6.

BAND LIMITED INTERPOLATION TO INCREASE SAMPLING RATE

The proposed procedure provides the corrected accelerogram at a sampling rate of 100 SPS. This sampling rate may not be sufficient for a particular application and in this situation the corrected accelerogram may be required to be interpolated to obtain the desired sampling rate. In *BLACS* this interpolation is done such that the frequency contents of the data do not change. This is achieved by performing the band pass interpolation which is done by inserting required number of zero in between the samples and performing low pass filtering with a cut off frequency equal to Nyquist frequency of initial sampling rate (50 SPS in this case). The details of this process is discussed by Basu *et al.* (1992).

To study the response of the entire scheme, a white noise at 200 SPS is generated. This white noise is used as uncorrected accelerogram to the scheme. The natural frequency of transducer is assumed to be 20 Hz and the damping is assumed to be 60% of critical. The corrected data at 100 SPS is band limited interpolated to get a

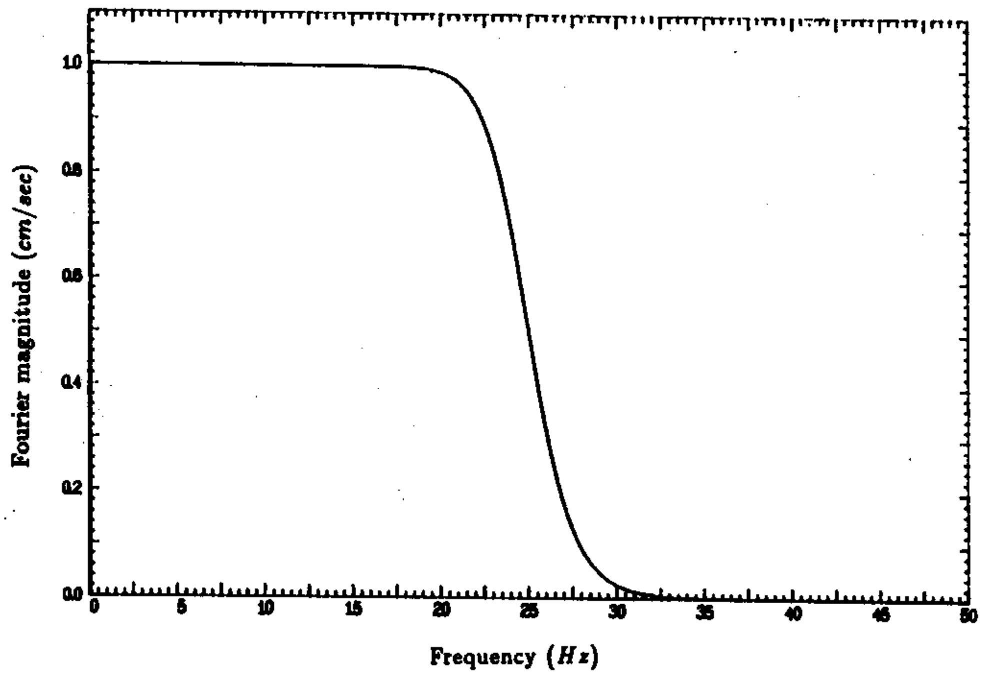

Figure 5: Response of proposed band pass filter to white noise input.

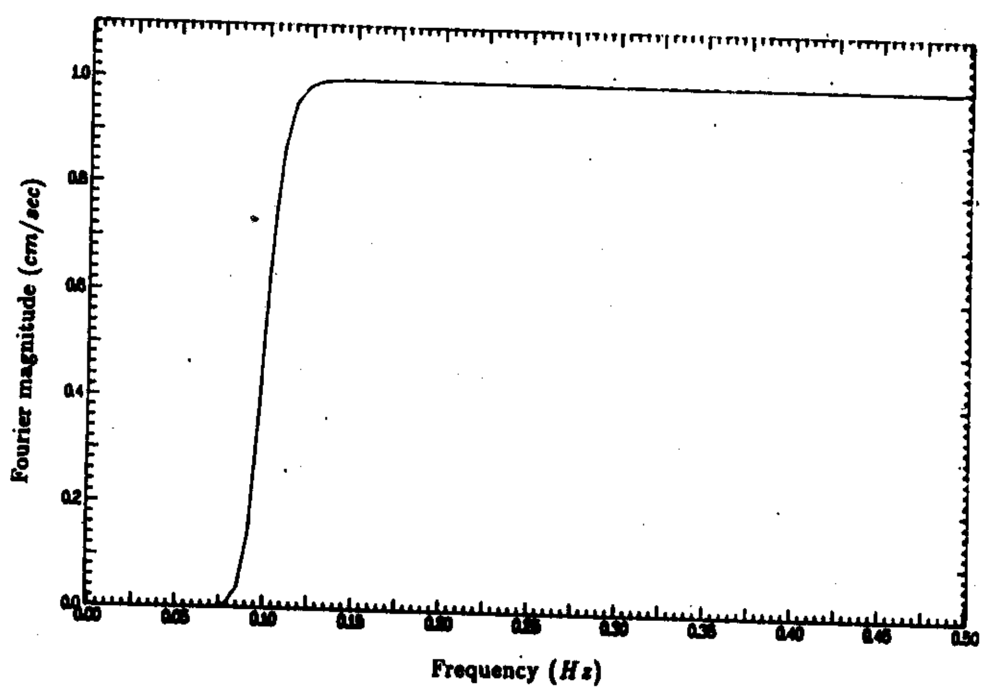

Figure 6: Zoomed Figure 5 to show response of high pass portion of band pass filter.

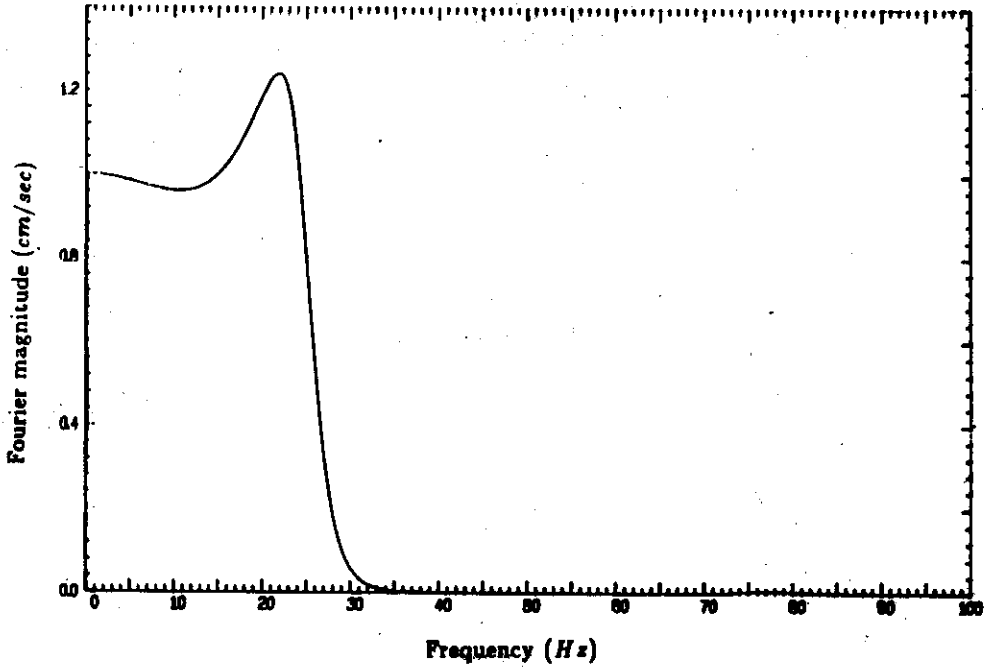

Figure 7: Response of the proposed correction scheme to white noise input.

sampling rate of 200 SPS. The Fourier transformation of this sequence is taken and Fig. 7 gives the Fourier magnitude plot of the response of the entire correction scheme to white noise.

INTEGRATION FOR VELOCITY AND DISPLACEMENT

Determination of ground velocity and ground displacement histories from the records of analog accelerographs will perhaps always remain questionable. This is due to the fact that some of the low frequency signals which predominantly effect the process of integration get overshadowed by the low frequency noise and, therefore, have to be removed through high pass filter. Also the initial conditions of zero displacement and zero velocity do not hold true for records of analog accelerographs which get triggered when the ground acceleration exceeds some threshold value. Nevertheless, integration of corrected accelerogram is performed with zero initial conditions to get the velocity and displacement histories. In the present scheme the integration is performed in the frequency domain.

Let $x_c(t)$ be the corrected accelerogram and let $X_c(\omega)$ be its Fourier transform and let $r(\omega)$ and $s(\omega)$ be the real part and imaginary parts respectivily of $X_c(\omega)$, then

$$X_v(\omega) = \frac{X_c(\omega)}{j\omega} \tag{20}$$

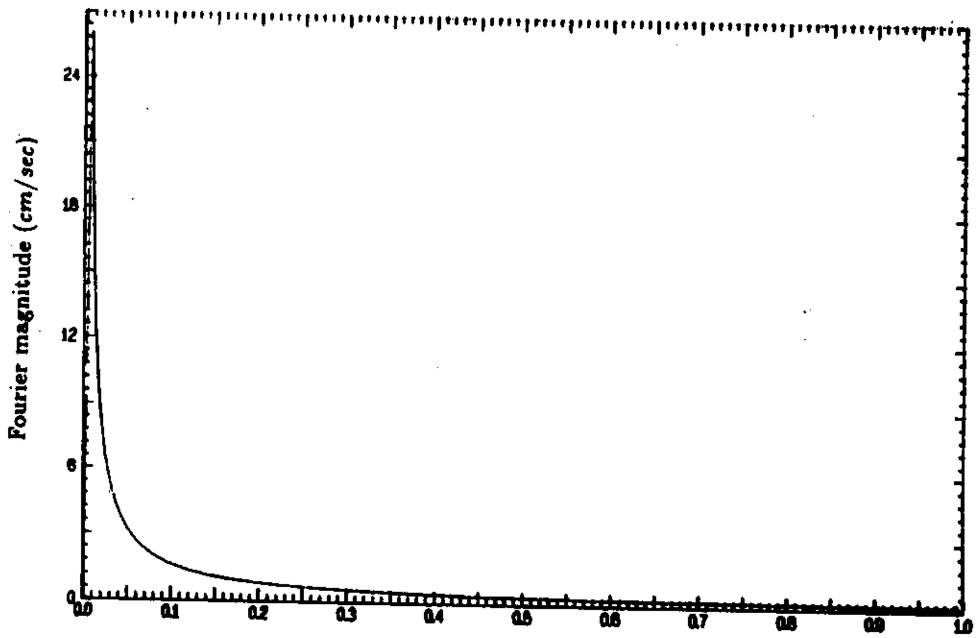

Figure 8: Response of integration process to white noise input.

where $X_v(\omega)$ is the Fourier transformation of velocity. In other words if $b(\omega)$ and $c(\omega)$ respectivily are the real and imaginary parts of $X_v(\omega)$ then, Eq. 20 means

$$b(\omega) = \frac{s(\omega)}{\omega} \tag{21}$$

and

$$c(\omega) = -\frac{r(\omega)}{\omega} \tag{22}$$

With Eqs. 21 and 22, the real and the imaginary parts of the Fourier transformation of the ground velocity are found which on inversion yields time history of ground velocity. To account for the zero initial condition, $b(\omega)$ and $c(\omega)$ are taken as zero for $\omega = 0$ (at the first bin frequency).

To check the performance of this integration process, a white noise of 100 SPS with 2048 data points is generated. This white noise is integrated with the above process. The Fourier transformation of this integrated signal is then taken. Figure 8 gives the Fourier magnitude plot of this integration process. The plot matches with the transfer function of an ideal integrator.

AN ILLUSTRATIVE EXAMPLE

This correction scheme has been used to process several uncorrected accelerogram and details of some of these are discussed in second part of this paper which deals with the comparison of *BLACS* with other schemes in frequency domain. However, the second part of this paper does not contain any figure of time history of corrected accelerogram and derived velocity and displacement obtained through *BLACS*. A sample of corrected accelerogram of Uttarkashi earthquake (Report no. 93-07 of Department of Earthquake Engineering, University of Roorkee) with derived velocity and displacement histories obtained from this scheme is presented in Fig. 9. In this figure the cut off frequency of high pass filter was taken as 0.05 *Hz* which is generally taken as standard with out consideration of inherent noise level of instrument. This figure shows improbable high values of velocity and displacement at the end of record. However, if noise level of RESA V on which this earthquake was recorded and noise to signal ratio for this earthquake is considered in selecting the appropriate cut off frequency (Kumar *et. al*, 1993) then for a signal to noise ratio of 10, cut off frequency for the high pass filter should be 0.5 *Hz*. Figure 10 gives time history of corrected acceleration, velocity and displacement of the same earthquake obtained through *BLACS*.

THE SOFTWARE

The program *BLACS* developed for this correction scheme is comprehensive and interactive in nature. It demands from the terminal all the input and output file names. The input file names demanded are the time mark, the fixed trace and the earthquake files which are made by the digitizers. The names of the output files which the computer programme generates are also taken through the terminal. These files are uncorrected accelerogram at nonuniform sampling rate, uncorrected accelerogram at 200 *SPS*, corrected accelerogram at 100 *SPS*, corrected accelerogram at the desired sampling rate, velocity history and displacement history. The program also demands from the terminal, the various parameters of the process like natural frequency of accelerometer, damping of accelerometer, value of λ of iteration for recovery of signal from nonuniform samples, maximum number of iterations, cutoff frequency of low pass and high pass filters ,the order of the filter and the number by which the sampling rate of the corrected accelerogram is desired to be increased. The options available with the computer program is to prepare a documentation file which prepares in the tabular form the acceleration, velocity and displacement histories with a header which specifies the details of earthquake, name of station, component orientation, cut off frequencies etc. The other options are preparation of response spectra and Fourier spectra of corrected accelerogram. A comprehensive terminal file can be prepared which provides all the inputs of the program in a sequential manner. This computer program *BLACS* is available to users on request.

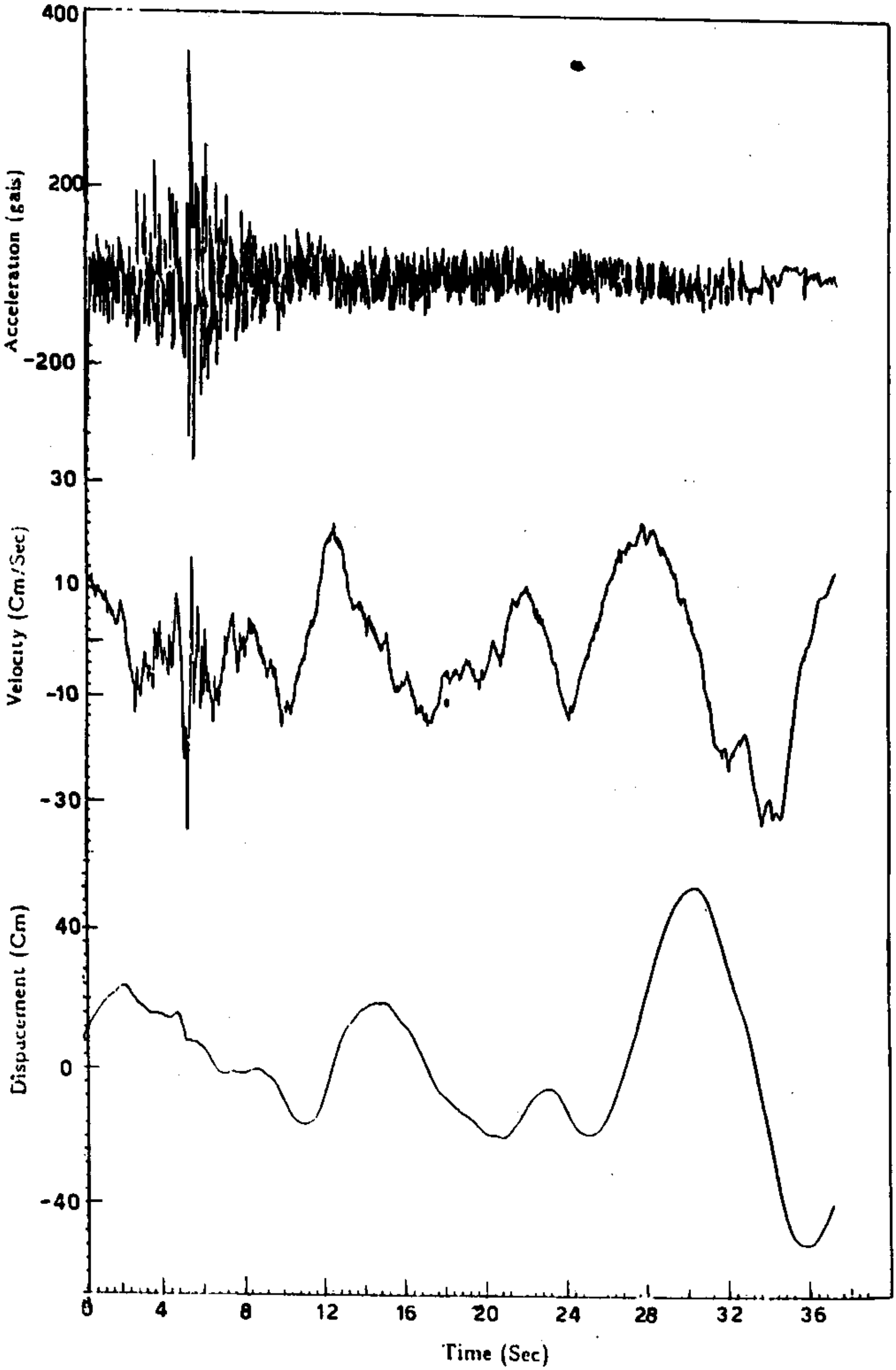

Figure 9: Time history of corrected acceleration, velocity and displacement of longitudinal component of Uttarkashi earthquake using BLACS with a cut off frequency of high pass filter as 0.05 Hz.

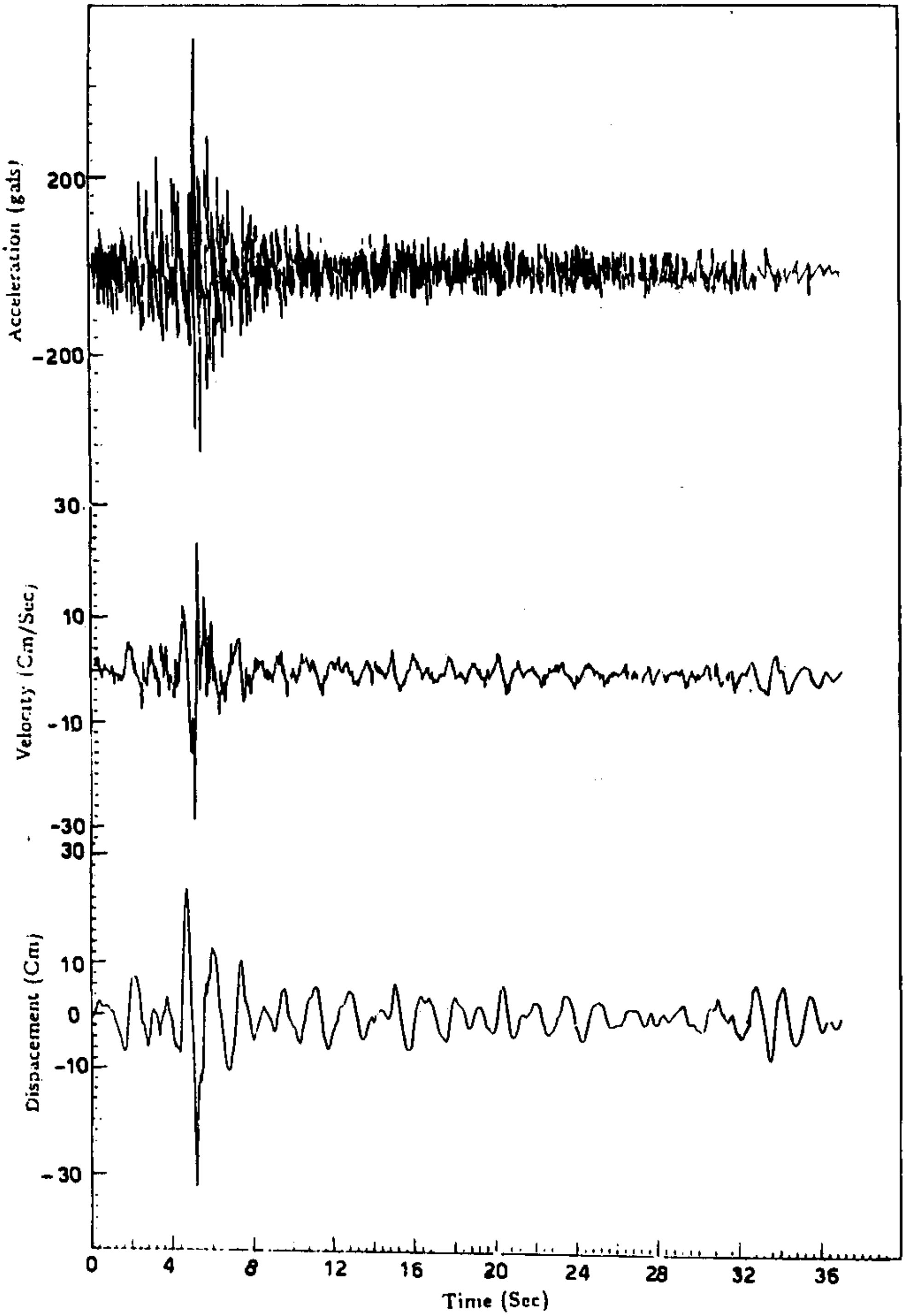

Figure 10: Time history of corrected acceleration, velocity and displacement of longitudinal component of Uttarkashi earthquake using BLACS with a cu⁺ off frequency of high pass filter as 0.5 Hz.

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