COMPUTIER **STUDIES** OF MICROSEISM **STATISTICS** WITH

APPLICATIONS **TO** PREDICTION **AND DETECTION**

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MITHDRAWN



## COMPUTER STUDIES OF **MICROSEISM** STATISTICS **WITI APPLICATIONS** TO PREDICTION **AND DETECTION**

*by*

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### ABSTRACT

Computational experiments have been performed on seismic data digitized from the records obtained **by** the Air Force during the Logan and Blanca underground nuclear **shots, by** Dr. Bruce Bogert in **New** Jersey **and by the** Wichita Mountain Seismic Observatory.

The experiments indicate that microseismic noise of about **.3** cps frequency is associated with the oceans but the higher **frequenoies** are not. Attempts to identify definite wave types, such as Rayleigh and Love waves, **and** to follow **wave** packets from station to station failed, but the failure illustrated the complexity of the microseisms **and points** out %he necessity of **a** statistical study.

For the statistical **studies** the microseisms were considered to be stochastic time series. It was found that the probability densitios of the amplitudes wre **Gaussian and** were not independent. Spectral analysis showed the typical microseism spectrum to have a **maximum** at about **.3** cps **and often other strong bands** at 1.4 and 2 **cps.**

The microseism time series are approximately stationary and can be described **as a** moving average operation. Thus they can be generated **by a convolution of a minimum phase** wavelet with **a** *white* light series. The wavelet is found for **typical** data **by** factorization of the power spectrum and the white light series is obtained **by** convolution of the inverse minimum phase wavelet with the noise data. Tests on the white light

series indicate that its probability density is approximately Gaussian and that it is approximately independent. Hence non-linear operators or filters are not particularly useful in microseism studies.

Cross correlation and cross spectra between different components of data at the same station, like components at different stations and array data have been computed. It was not possible to identify individual wave types or directions of travel with any degree of certainty.

Prediction studies of microseisms have been done to try to improve the signal to noise ratio during the first motion interval. The mean squared error technique and the spectrum factorization technique have been used. The spectrum factorization is found to be superior because long operators can be more readily obtained. However, one can predict at best about **50%** of the energy which is not sufficient to produce a significant improvement in the signal to noise ratio. Indications are that other prediction techniques will not give much better results.

Artificial microseisms generated **by** convolution of a typical microseism wavelet with Gaussian white has been used in a computer simulation of a detection system. The system is an energy detector which detects events in microseismic noise. The system is studied in terms of false alarm rate and failure to detect rate. Overall system effectiveness is given in terms of false alarms per hour as function of signal to noise ratio for a **95%** probability of detection success. The system characteristics are found to be essentially invariant when the inputs are band pass filtered. The simple band pass filter can in some cases give significant signal to noise ratio improvement.

Details of the statistical tests and computer programs are given along with an approximate solution to a non-linear water wave problem related to microseism generation. The solution, which uses DeVorkin's representation scheme, is for arbitrary initial conditions and shows that sum and difference frequencies of all the frequencies present initially will be generated.

Thesis Supervisor: Stephen M. Simpson, Jr. Title: Associate Professor of Geophysics

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The data was digitized with the aid **of** Wolf Research and Development Corporation and Research Calculations. The computation was done in part at the M.I.T. Computation Center with the help and cooperation of Michael Saxton of the IBM Liasion Office and in part at the M.I.T. Cooperative Computing Laboratory with the valuable assistance of Anthony Sacco.

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#### INTRODUCTION

#### Need to Study Noise

The disarmament talks at Geneva and the need for a surveilance network to detect and report the testing of nuclear devices, particularly underground testing, have put new emphasis on the field of Seismology. Government support in this area has made possible much research into the nature of seismic disturbances and instrumentation for detecting them. The present thesis was supported **by** the Advanced Research Projects Agency under the Vela Uniform Project contract **AF 19(604)7378.** The contract covers the digitization of the paper records from the Logan and Blanca shots of the **1958** Hardtack series, investigation of ways to improve the signal to noise ratio, particularly in the first motion interval, and investigation of the properties of bomb and earthquake signals.

### Definition of Microseisms

Essential to the problem of signal detection and signal to noise ratio improvement is an understanding of the natures of both the signal and the noise. This thesis will deal mainly with the properties of the noise. **A** definition of what is meant **by** noise is necessary since in many cases what is noise to one man is signal to another. In the context of this thesis any ground motion not associated with definite bomb or earthquake signals, motion which is present at all times, will be considered noise and will be called microseisms or microseismic noise.

The study of microseisms dates back about **100** years to the pendulum measurements of an Italian monk, Bertelli (Haq, 1954). Only very

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qualitative conclusions which generalized the data could be made, but it was obvious from study of Bertelli and others that the surface of the earth was in a state of oscillation. This "sea" of elastic waves came under the scrutiny of other observers who were interested in the causes of the disturbances. Wiechert (1905) suggested that microseisms were generated by the impact of surf on a steep coast. Gutenburg (1912) noted a correlation of microseisms with 4 to 8 second periods with surf and wind direction. Ramirez (1940) studied the physical properties of microseismic waves, the velocity, direction of travel and particle motion, with a tripartate or triangular arrangement of three component instruments. He found that the properties of these waves were fairly consistent with those of Rayleigh and Love waves.

### Sources of Microseisms

Observers noted that the microseisms and sea waves seemed to be connected, and, in some cases, the periods of the sea waves were twice the period of the microseisms. However, the idea that sea waves produced microseisms was hard to justify theoretically since pressure variations due to travelling water waves die out exponentually with depth and are nearly zero within a wave length. Miche (1944) showed that there is a pressure fluctuation under a standing wave which is unattenuated with depth (for incompressible fluids), and its frequency is twice that of the sea wave. Longuet-Higgins (1950) realized that this was what was needed to explain the observations. He also showed that the mechanism could account for the energy of the observed microseisms. The presence of an unattenuated double frequency variation is demonstrated by Longuet-

Higgins in a small parameter expansion approximation to the solution of the non-linear equations for the pressure variations at the bottom of a layer of water with a rigid lower boundary and a standing wave on the top. Another method of approximation for this type of problem using a representation scheme for the solution of non-linear equations worked out **by** DeVorkin **(1963)** is given in Appendix **A.** It illustrates that the sum and difference frequencies of all frequencies present initially are expedted to develop.

The microseisms with periods from 4 to 12 seconds are generally attributed to ocean waves and recourse to the theory of Longuet-Higgins can be made for their explanation although there is still controversy on the matter. The data which has been used in this thesis was recorded with a Benioff short period instrument so that only the shortest period oceanic microseisms come through. Microseisms of higher frequency than the oceanic band are usually attributed to wind and meteorological factors or are thought to be cultural noise. Typical noise sources are swaying trees and buildings, storms, city traffic, heavy machinery, power plants, trains etc.

This brief allusion to the history of the study of microseisms does not give a feeling for the enormous amount of work which has been done in this area. (See Haq, 1954, for a fuller account and references.) **A** great deal of the work has been concerned with microseism generation mechanisms, surface wave propagation and particle motion, and studies of the direction of propagations and their relation to storms. Nearly all of these studies consider microseisms as a signal. This thesis for the most

part considers microseisms as noise. The main object is to treat the microseisms from a statistical point of view and try to describe them so that something can be done about them rather than with them. To this goal, the tools of statistical analysis have been brought forward and applied with the aid of high speed digital computers.

We shall see that a few examples which treat the microseisms as signals will suffice to point out the nbed for a more general description of the noise. It is obvious that that time series analysis can be applied to the study of microseisms, but stronger and more useful statements can be made about the time series if it can be shown that they are stationary or, better still, ergodic. We must therefore test the microseisms to see if they **fall** into one or more of these special categories of time series. Spectral analysis, probability studies and independence tests are some of the techniques which aid in the classification of microseisms.

The proper mathematichl description of microseisms can also be the key to the optimum prediction problem, and will permit the study of the predictability of microseisms. We shall see that prediction can be used in some cases to reduce the noise level and therefore, if a signalis also present, improve the signal to noise ratio. The amount of improvement is of course dependent on the predictability of the noise.

**A** good mathematical model of microseismic noise will also permit us to generate the noise artificially. This artificial noise is extremely useful when long sections of continuous noise are required, and is therefore necessary when we simulate **by** computer a system to detect events in microseismic noise.

**Outline** 

The thesis is divided into four chapters. The first deals with the basic statistics of the data on which the present studies are based. It includes a description of the data and how it was recorded as well as amplitude studies, auto and cross spectra, empirical probability density functions, and a mathematical model for noise generation.

Chapter two discusses the prediction of the noise **by** different methods and then applies this to the problem of the determination of the direction of first motion of a signal in the noise. Improvement with non-linear predictors is also considered.

In chapter three an automatic system for the detection of signals in microseismic noise is proposed and the results of a computer simulation of this system are given in terms of detection probabilities and false alarm rates for filtered and unfiltered inputs.

Chapter four is a summary which restates the major conclusions.

Details of some analyses and the computer programs used are left for the Appendices.

### **1. BASIC** STATISTICAL STUDIES

# 1.1 Empirical Data

# Data Sources **-** Noise before and Noise after Events

The data which forms the basis for most of the computational studies described in this thesis are the seismic records of the Logan (5 KT) and Blanca (19KT) underground nuclear shots of the 1958 Hardtack series (Romney, 1959). These were recorded by the **U. S.** Air Force at **28** temporary stations set up across the United States as shown in Figure **1.1.1.** The instruments used were short period Benioffs with galvanometer periods  $(T_1)$  of .20 seconds. Most stations were equipped with a vertical instrument (up-down) and two horizontals,a "toward-away" and a "right-left". These designations are with respect to an observer standing at the shot point looking at the station. The vertical and horizontal instrument responses are the same and are shown in Figures 1.1.2 and **1.1.3** (Geotechnical Corp., **1961).** The paper records from these shots were provided by the Air Force and were digitized at 20 samples per second. In no case were the paper records for an entire drum revolution provided so that the greatest time interval of continuous record available was on the order of a few minutes. For this reason the noise records which have been digitized are labeled "Noise Before" and "Noise After" with the appropriate shot, distance from shot and component. Noise before refers to the trace on the paper record which is just above the signal trace, and is therefore one drum revolution

time before the shot. Noise after is the trace just below the signal trace. **A** copy of one of the original paper records which was digitized is shown in Figure 1.1.4, and a plot of the corresponding digitized record is shown in Figures **1.1.5** to **1.1.7.** Figures **1.1.5** to **1.1.7** have been plotted **by** computer program using the oscilloscope attached to the IBM **7090** computer at the M.I.T. Computation Center. These graphs, and many of the others appearing in later sections, have been plotted as histograms. In several cases, particularly the spectral computations, the values plotted are averages or estimates over some range so that there is no justification for interpolation and the histogram is the predered method of presentation.

### Logan and Blanca Digitization Procedure

The records were broken up into sections and each section was digitized separately. This procedure can lead to some error since each section could have a linear trend. This was compensated for **by** removing the best fitting (in the least squares sense) segmented line from the entire record, where each segment is the length of a section.

The digitization accuracy is good to a few percent, and the gain values supplied with the original records are quite good, but the actual ground motion values may be off by as much as **15** percent.

Other digitized data has been provided **by** Dr. Bruce Bogert of the Bell Telephone Laboratories, who has a short period vertical Benioff' at Cherry Hill Park, New Jersey, and **by** United Electro Dynamics, Inc., who have digitized the records from the WMSO station in Oklahoma. Dr. Bogert's Benioff has a response similar to that of the Hardtack instruments, but its

low frequency cut **off** is somewhat higher (Bogert, **1961),** Figure 1.1.8. The WMSO station is a linear array of vertical Benioffs with the same response as the Hardtack instruments.

**A** list of our record numbers appropos to this thesis and the event and station to which they correspond, is given in Table 1.1.1.

# TABLE 1.1.1









Figure 1.1.2



 $\mathcal{A}$ 

 $\cdot$ LOGAN<br>1902 Km TEarth Up<br>Gam 173K TEarth Up  $6643.492$ **Same Company** SAN ANSET STAATSLIKT AF NOONTSKANSING IN DIE TRAFFICIALISTE AFRIKANSE KAN DIE TRAFFICIALISTE ANDERS KONSTANTIN nanovani  $05.08149.2$  $LocAN$ <br>1902 Km 1 Toward  $Gain 200k$  $\int$  LET manimuminum LOGAN<br>1902 Km (Left<br>Gain 148K (Left  $06:03:49.2$ 

Figure 1.1.4



 $FIG. 1.1.5$ 



 $F[G. 1.1.6]$ 



FIG. 1.1.7



**Figure 1.1.8**

# 1.2 Elementary Properties

We shall briefly consider the microseisms as a signal in a few somewhat naive computational experiments which will suffice to make apparent the need for a more general approach to the study of microseisms which can be provided **by** statistical techniques.

The first experiment, which is concerned with microseism amplitudes, has some bearing on microseism sources and the results are in agreement with those obtained **by** others. The second set of experiments deals with the identification of wave types, specifically Rayleigh and Love waves, in the microseisms. As we shall see this set of experiments failed badly because of the simplicity of the model which is used and the complexity of the microseisms themselves.

### Microseism Amplitude Studies

Some studies have been made on the amplitudes at two frequencies of the noise from the Logan and Blanca records to determine the change in amplitude with distance from an ocean. If the microseisms, at the frequencies in question, are of oceanic origin, there should be a definite decrease in amplitude with distance from the coast. The frequencies and amplitudes were estimated directly from the paper records. The approximate frequency values were obtained by counting peaks over a minute or more of record. On almost all the records, the noise appeared to have two distinct frequencies, one at about **.3** cycles/second, and the other near 2 cycles/second. Approximate peak amplitudes were measured on the records and averaged over several cycles of the frequency of interest. An attempt was made to choose an average noise trace before the shot.

A plot was then made of amplitude versus distance from the Atlantic or Pacific coast (whichever was closer) for both frequencies. These graphs appear in Figures 1.2.1 and 1.2.2 for Logan and Blanca respectively.

We can see from these figures that for low frequency the noise decreases for inland stations, but for the higher frequency there is no systematic trend. The increase in amplitude of the low frequency component at about 1400 km from a coast may be due to microseisms from the Gulf of Mexico. These rather rough quantitative results are as expected, since the low frequencies are usually assumed to be caused by ocean waves and the high frequencies are attributed to local sources, and are not correlated with the distance from the coast.

It is interesting to note that the rough computation of the frequencies involved is supported by detailed spectral analysis. Figures 1.3.6 to 1.3.9 show spectra of some of the noise and it can be seen that the important frequencies are at about .3 cps, 1.4 cps and 2 cps for the Logan and Blanca records.

#### Rayleigh and Love Wave Experiments

Much of the energy in microseismic noise has been attributed to surface waves of the Rayleigh and Love wave types. Studies by several observers mentioned in the introduction have indicated the presence of these waves in the 4 to 8 second period range. The spectrum of noise from Logan, Blanca and Cherry Hill Park records which appear in Figures 1.3.6 to 1.3.9 show spectral lines with most of the energy concentrated in fairly narrow bands. The low frequency peak, as was mentioned before, is a bit artificial, since it is the high frequency end of the oceanic

microseism band with the low end cut off **by** the Benioff response. We might well suppose that this peak is composed of Rayleigh waves. The higher frequency lines may also be Rayleigh waves but of a non-oceanic origin. The Cherry Hill Park records in Figure **1.3.9** are remarkably similar, with rather narrow bands, even though they were taken three months apart, and one would like to investigate the important frequencies to identify wave types. Unfortunately, there are no horizontal recordings available and thus no study of this nature can be done. However, the Logan and Blanca records are three component and some attempt has been made at wave type identification. The spectra of these records, Figures **1.3.6** to **1.3.9,** show in general more energy in the horizontal components at high frequency than in the vertical component. This suggests that the higher frequency noise, 1.4 cps and 2 cps, may be Love waves, and the possibility that the lower frequency energy is due to oceanic microseisms is still present.

Rayleigh waves are a special combination of P waves and **S-V** waves which confine all particle motion to a plane defined **by** the vertical and the direction of travel of the waves. For a single frequency the partical motion is retrograde elliptical. Assuming, therefore, that we have a single Rayleigh wave of a single frequency, we can resolve the horizontal components of motion into a new coordinate system which is rotated with respect to the original seismometer coordinate systems such that all horizontal motion is along one axis, the X" axis. This axis then determines the direction of travel of the wave, but not the sense of the direction. The sense can be determined from the resolved horizontal, X", and the vertical, Z", components. Since the partical motion

is retrograde elliptical, X" must lead Z" by **900** for the wave to be travelling in the positive X" direction. A plot of X" against Z" should be an ellipse with its X" intercept almost 2/3 of its Z" intercept.

Records 2000, 1002 and 1004, the noise before the Logan shot 1902 km from the shot point, form a three component set and therefore can be checked in the manner described for a Rayleigh wave component. All three records were band pass filtered with a filter of width .08 cps centered at .255 cps. This frequency corresponds to the maximum of the spectrum and is possibly attributable to Rayleigh waves from oceanic sources. The two horizontal components were plotted against each other and a line fitted to the plot. The plot was fairly scattered so that the fit of the line was quite poor. The horizontal to vertical component power ratio after rotation was only 5 which is not correct for Rayleigh waves. If the plot fell exactly on a straight line the ratio after rotation would be zero. The indication is that the plot was not even close to a straight line. The resolved horizontal component was then plotted against the vertical and an ellipse was fitted to the resulting curve. This plot was the best fitting ellipse superposed is shown in Figure 1.2.7. The ellipse in this figure is a very poor fit and it is not possible to reconcile these results with the single Rayleigh wave hypothesis. This does not mean that the low frequency peaks are not Rayleigh waves. Presence of two or more Rayleigh waves from different sources could explain the lack of a linear relationship between the horizontal components and the poorly fitting ellipse to the horizontal versus vertical plot. We might note, however, that some of the motions shown in Figure 4.2.1 are relatively elliptical, but with

tilted axes. Examination. of the spectra (Figures **1.3.6** to 1.3.8) shows relatively more power in the vertical at **.255** cps than we would expect on the Rayleigh wave hypothesis, but this could be explained **by** a mismatch of seismometer characteristics.

**A** test for the presence of Love waves was also performed on this data. The peak at about 2 cps was of interest here, since there was relatively more power in the horizontal than in the vertical. For a single Love wave we would again expect that a plot of the horizontal components would fall on a straight line. This was not the case, however, for a band width of about **.08** cps centered at **2.05** cps. It is most probable that either Love or Rayleigh waves from a single source do not occur, or the band width used is too wide to see them. Cross correlation experiments could be most useful here, since the equivalent band width is the Daniell window width and the phase at each window width may be easily checked. For Rayleigh waves, we expect the horizontal to be in phase, but 90° out of phase with the vertical. For Love waves the horizontal should again be in phase, but there should be very little energy in the vertical component.

The failure of these two experiments does not eliminate the possibility of the existence of Rayleigh and Love waves at the frequencies considered, but it does illustrate the complicated nature of the noise. The suggestion is, therefore, that the structure of the microseisms is too complex to be handled by simple deterministic models. Rather than introduce more complicated models which require an enormous amount of labor to fit to the data, we shall consider the microseisms as stochastic time series and treat them from the statistical point of view.

### Apparent Stationarity

The majority of the results of time series analysis are applicable to stationary time series, that is, series whose probability densities are not dependent on absolute time. If in a time series the probability,  $\mathbb{R}_{\zeta_1}(X_i;t_i)$   $dx_i$ , that  $\xi_i$  is in the interval  $(X_i,X_i+\epsilon x_i)$  at time  $t_i$ is the same for all  $t$ , and if the probability  $P_{\underset{1}{\bigcup}\underset{2}{\bigcup}}$  ( $x_{1},x_{2},t_{1},t_{2}$ ) that at time  $t_i$ ,  $\xi_i$  is in the interval  $(x_i, x_i+\epsilon x_i)$  and at time  $t_2$ ,  $\xi_1$ is in the interval  $(x_{\lambda},x_{\lambda}+dx_{\lambda})$  is dependent only on the time separation  $\tau = t_1 - t_1$  and not on absolute time, the time series is said to be wide sense stationary. If all higher densities  $P_{\xi_1\xi_2\cdots\xi_n}(x_1,x_2\cdots x_n)$  $t_{1}, t_{2}, \ldots t_{n}$  are also independent of absolute time and dependent only on  $\tau_{\kappa} = t_i - t_i$  the series is strictly stationary.

It is obvious that microseism records are not stationary over long periods of time since microseism activity is strongly influenced by meteorological conditions. Over short periods of time, however, when there have been no great changes in the generating mechanisms for microseisms, the records can be considered stationary. For our purposes we need only be concerned with stationarity over the few hours necessary to record the shot signal and noise before and after the signal. We now consider an ensemble or group of time series lined up one beneath the other each with the same first and second probability densities. We arbitrarily label time on these series so that a vertical line strikes each time series at the same time. The ensemble can be constructed by breaking up a long time series into smaller pieces **and** considering each piece as a member of the ensemble. In the case of microseismic noise, the noise before and the noise after the event can be considered as two members of the ensemble. We wish then to see if the probability densities

are approximately the same for these ensemble members. We can do this computing directly the probability densities, but this becomes a lengthy process for the second density,  $P_{\xi_1\xi_2}$  ( $X_1, X_2$ ;  $t_1, t_2$ ) and it is worse for the higher densities. If we are only interested in wide sense stationarity we can consider time and ensemble averages and, assuming that the ensemble is ergodic, equate these averages. The ensemble average of  $\xi_1$  at time  $t_1$  and  $\xi_2$  at time  $t_2$  is

$$
A_{\nu e} = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} x_{1} x_{2} P_{\xi_{1} \xi_{2}}(x_{1}, x_{2}; \tau) dx_{1} dx_{2}, \xi_{2} \tau_{2} \tau_{1}
$$

The time average is

$$
Ave = \lim_{T \to \infty} \frac{1}{2T} \int_{-T}^{T} f(t) f(t + \tau) dt
$$

We note that the time average is the autocorrelation and that the Fourier transform of the autocorrelation is the power density spectrum (see section **1.3).** Hence, under the ergodic hypothesis, the constancy of the spectral density in time reflects the wide-sense stationarity of the time series. Spectral density computations have been performed on the noise before and noise after the shot and the results are shown in Figures **1.3.6** to **1.3.8.** One can easily see that the general character of the spectrum does not change much over a period of time representing two drum revolutions of the Benioff. This strongly suggests that the microseisms are, for our purposes wide-sense stationary.

### Mean and Variance

Time series analysis simplifies to some extent if the series have zero mean and unit variance. The digitized records had the best least squares fitting segmented mean line removed, but this does not guarantee that the mean is zero. The mean is, however, quite small and can usually be considered zero. It can easily be computed and subtracted off if necessary. The variance of the records is not unity and no scaling has been done to make it so.

## Amplitude Distribution and Normality Test

The amplitude distribution of the records can easily be computed and, given the mean and standard deviation (square root of the variance), the corresponding normal distribution can be found and compared with the empirical amplitude distribution. Appendix B gives a flow graph of the necessary steps in tie comparison of the distributions and the programs necessary. Appendix G contains listings of the programs. The comparison is done by finding the values along the x axis which divide the appropriate normal density (given mean and standard deviation) into sections of equal area (equal probability). A count is then made of the number of amplitude values which fall into each section. The chi square comparison measure is then

$$
\chi^2 = \sum_{i=1}^L \frac{(N_i - pN)^2}{pN}
$$

where there are L sections and N amplitude data points,  $P = 1/L$ , and  $N_i$ is the number of points which fall in the  $\int$ <sup>+</sup> $\uparrow$ *h* section. There are  $\downarrow$ <sup>-</sup>3

degrees of freedom since the mean and standard deviation are used to determine the appropriate Gaussian. The chi square measure thus defined is chi square distributed and its expected value depends only on (Cramer, 1946). The probability  $P(\lambda^{\lambda})$  of exceeding  $\lambda^{\lambda}$  is the quantity of importance in comparison. Acceptance regions for  $x^2$  are generally set so that  $P(X)$ <sup>2</sup>  $\geq$  .1 or .01. Comparisons were made between empirical and normal probability densities for all the Logan and Blanca noise records listed in Table 1.1.1. The chi square test was used as a measure of goodness of fit and the results are shown in Table 1.4.1 in section 1.4. The probability of exceeding  $x^2$  varies considerably and for the records shown only six or seven can be considered normally distributed for this test. Figures 1.2.3 and 1.2.4 show some of the empirical frequency ratio plots and Figures 1.2.5 and 1.2.6 show typical computer output from the normalcy and independence tests. It can be seen from these figures that even though some of the densities fail the  $x^2$  test, they look fairly Gaussian and to a rough approximation may be considered normal.

(Note: If the alternate method of test for normality which is given in section 1.4 is used, all records are found to be Gaussian.)

The independence tests are discussed further in section 1.4 and in Appendix C. It is sufficient to say here that the amplitudes are not independent.







Figure **1.2.3** Frequency Ratios **of** Microseism Amplitudes






ANALYSIS OF AMPLITUDE DISTRIBUTION FOR RECORD 1005 COMPARISON OF ACTUAL DISTRIBUTION AND NORMAL DISTRIBUTION

> NUMBER OF RANGES= 57 LENGTH OF SERIES= 3321 DEGREES OF FREEDOM= 54 MEAN OF SERIES= - 0.22500189E-05 STANDARD DEVIATION= 0.14274400E-02

HIGHER CENTRAL MOMENTS THIRD MOMENT= - 0.19685886E-09 FOURTH MOMENT= 0.12106580E-10 FIFTH MOMENT=  $-0.12533012E-14$ SIXTH MOMENT= 0.11494952E-15

EXPECTED COUNT= 58.2632

 $CHI-SQUARE =$ 0.62046965E 02 0.21316E-00 PROBABILITY OF EXCEEDING CHI-SQUARE=

## POKER COUNT TEST RESULTS



MEAN SQUARE CONTINGENCY= 0.27838460E 01

DEPENDENCY MEASURE= 0.30931623E-00

#### PROBABILITY DISTRIBUTION

NUMBER OF VALUES IN EACH OF 100 EQUALLY SPACED RANGES FROM -0.47553504E-02 TO 0.45647645E-02. 3321 VALUES IN ALL.



ANALYSIS OF AMPLITUDE **DISTRIBUTION** FOR RECORD **1026** COMPARISON OF **ACTUAL** DISTRIBUTION **AND** NORMAL **DISTRIBUTION**

> **NUMBER** OF **RANGES= 59 LENGTH** OF **SERIES= 3581 DEGREES** OF FREEDOM= **56 MEAN** OF **SERIES= -0.37916552E-07 STANDARD DEVIATION=** 0.13271835E-02

HIGHER CENTRAL **MOMENTS** THIRD **MOMENT=** -0.84812047E-10 FOURTH MOMENT= 0.97164132E-11 FIFTH **MOMENT=** -0.29763772E-14 SIXTH **MOMENT= 0.86117256E-16**

**EXPECTED COUNT=** 60.6949

**CHI-SQUARE\*** 010001674E **03** PROBABILITY OF EXCEEDING CHI-SQUARE= **0.15617E-03**

# POKER **COUNT TEST RESULTS**



**MEAN SQUARE CONTINGENCY= 0O23302333E 01**

DEPENDENCY **MEASURE= 0 .25891481E-00**

## PROBABILITY DISTRIBUTION

NUMBER OF **VALUES** IN **EACH** OF **100** EQUALLY SPACED RANGES FROM **-0o48722361E-02** TO **0.41697387E-02\* 3581 VALUES IN ALL.**





Horizontal Motion Figure 1.2.7 Results of Rayleigh Wave Experiment on Records 1000, 1002 and 1004 with Best Fitting Ellipse.

# **1.3** Correlation and Spectral Properties

# Description of Random Functions **-** Correlation and Spectrum

The description of the spectrum of a random function, such as microseismic noise as recorded on a seismogram, cannot be adequately done **by** simple Fourier transformation since the Fourier transform specifies the phase spectrum and immediately particularizes the function thus setting it aside from all the other possible realizations of the random process. In order to treat all the members of the ensemble simultaneously we must make use of the Wiener theorem for autocorrelation. The autocorrelation,  $\varphi(\tau)$ , of a continuous time function  $f(t)$ is defined as

$$
\varphi_{(\tau)} = f_{\tau} \ddot{r}_{\infty} + \int_{\tau}^{\tau} f(t) f(t + \tau) dt
$$

With a change of variables  $\forall z \uparrow \uparrow \uparrow$  we can see that  $\forall (z) = \forall (-\tau)$ . The Wiener theorem then states that the power density spectrum  $\Phi(\omega)$  of  $f(t)$  is the cosine transform of  $\mathcal{V}(t)$  (Lee, 1960).

$$
\Phi(\omega) = \frac{1}{2\pi} \int_{-\infty}^{\infty} \varphi(\tau) \cos \omega \tau \, d\tau
$$

We see that the autocorrelation has the effect of bringing all the phases down to zero thus throwing away the phase information which pins down a particular member of the ensemble.

The continuous infinite theory has its counterpart in discrete finite time, but with some modification and some problems.

## Digitization and Aliasing

Digitization or division into discrete time puts some restriction on the description in the frequency domain. One must pay the price for throwing away the information between the digitized points and that price, as specified **by** the sampling theorem, is that one can only see frequencies which are less than or equal to half the sampling rate. If there are  $h$  samples per second we can only distinguish up to  $\frac{n}{2}$ cycles per second, the Nyquist frequency, which corresponds to a radian frequency of  $W \times \mathbb{T}$ . If the data actually contain a frequency higher than  $n/2$  cps., say  $n/2 + \Delta$ , this frequency will be folded down to  $h/2 - \Delta$ , since  $\cos(\pi + \epsilon) = \cos(\pi - \epsilon)$ , and this process is called aliasing. Thus if there are frequencies present higher than  $h/2$  cps, the spectral estimate at frequency  $f_1$  ( $o \le f \le h/2$ )<sub>is made up of fre-</sub> quencies  $f(x) = \frac{2(n/2)t}{t}, \frac{4(n/2)t}{t}, \dots, M(n/2)t$ , *M C*  $e^{i(n/2)t}$ and the spectrum loses meaning. One can avoid this problem **by** sampling often enough to include all frequencies or **by** low pass filtering before digitization.

## Spectral Estimation - Daniell Window and Variance of Estimate

The fact that the data is known for a finite length of time requires an assumption about the data outside of the interval in which it is known since the autocorrelation  $\forall$  ( $\tau$ ) involves this time. One usually assumes that the data is zero outside this interval and the autocorrelation must therefore go to zero when  $\tau$  equals the interval length. This is the complete transient (Wiener) autocorrelation

$$
\psi_{i}(r) = \frac{1}{N} \sum_{i=0}^{N-1} X_{i} X_{i} + r \ , \quad T = 0, \pm 1, \pm 2,..., \pm (N-1)
$$

where there are  $N$  data points,  $X_{1}$ ,  $X_{2}$ ,  $\ldots X_{N}$  . Some methods of estimating the autocorrelation such as the Tukey estimation try to compensate for the fact that the data is zero outside  $\vec{l} : \vec{l}$ , ...  $\lambda \vec{l}$  by adding weighting factors

$$
\varphi_{(\tau)} = \frac{1}{N-|\tau|} \sum_{i=0}^{M} X_i X_{i+\tau}, \qquad \tau = 0, \ldots \pm m
$$

where  $M$  is less than  $M$  (e.g.  $M \cdot N / 5$ ). The higher lag terms **(** T large) are thus given more weight to compensate for the smaller number of terms in the summation. This will, of course, result in a biased estimate.

In any case the computed spectrum,  $\Phi_c(\omega)$ , is an estimate of the true  $\vec{\Phi}(\omega)$  and can be though of as a convolution of some weighing function  $\mathsf{W}(\omega)$  with the true spectrum

$$
\Phi_c(\omega) = \Phi_{\tau}(\omega) * \sqrt{\omega}
$$

where the asterisk denotes convolution.  $W(\omega)$  is then called the spectral window (Blackman and Tukey, **1958).** Ideally the spectral window is rectangular and the convolution process will then move it along the true spectrum and the estimate at  $w_{\kappa}$ ,  $\Phi_{c}(\omega_{\kappa})$  will be an unweighted average of the true spectrum  $\Phi_{\tau}(\omega)$  from  $W_{\kappa}+\mu$  to  $W_{\kappa}+\mu$  where  $2h$  is the window width. Since convolution in one domain is multiplication in the other, the Fourier transform of  $\Phi_{\tau}$  (w)  $*$   $\mathsf{W}(\omega)$ is  $\int_{\tau}(\tau) W(\tau)$  where  $\int_{\tau}(\tau)$  is the true autocorrelation.

The spectral estimate which has been used to compute the spectra and cross spectra shown in this thesis is the Daniell estimate. The Daniell method uses the complete transient (Wiener) autocorrelation of the time function  $X_t$ ,  $t = 1$ ,  $\ldots$  **N** 

$$
\varphi_{(\tau)}: \frac{1}{N} \sum_{t=1}^{N-1} x_t x_{t+\tau}, \quad \tau = 0, \pm 1, ... \pm (N-1)
$$

The Daniell spectral estimate  $\Phi_{\mathfrak{d}}(\omega)$  is then

$$
\Phi_{\mathbf{b}}(\mathbf{w}) = \frac{1}{2\pi} \sum_{\tau=-(\mathbf{w}-i)}^{\mathbf{w}-1} \varphi(\tau) \underset{\pi \in \mathcal{T}/m}{\xrightarrow{\text{SIN}}} \cos \omega \tau \quad (1.3.1)
$$

where  $\frac{S_{i,n}(\pi T/m)}{\pi \gamma i_m}$  is the Daniell weighting function. We note that the spectral window is not simply the Fourier transform

of the Daniell weight since  $\psi(x)$  is not the true autocorrelation. We can, however, compute the spectral window if we choose a time function  $X_t$ for which we know  $\oint_{\tau}$  ( $\omega$ ) (Simpson et al, 1961b). If the time function  $X_t$  is N points of a sine wave sin  $\omega_r t$  we know that  $\Phi_{\tau}(\omega)$ is a delta function  $\int (\omega_{\nu})$  so that the spectral estimate becomes

$$
\Phi_{\mathbf{D}}(\omega) = \Phi_{\mathbf{T}}(\omega) * \mathbf{W}(\omega)
$$
\n
$$
\Phi_{\mathbf{D}}(\omega) = \mathbf{S}(\omega_{\mathbf{r}}) * \mathbf{W}(\omega) = \mathbf{W}(\omega_{\mathbf{r}}\omega_{\mathbf{r}})
$$

Hence we compute the transient autocorrelation  $\mathcal{P}_{i}$  of trom the  $N$  points

of the sine wave, weight this with the Daniell weighting function and take the cosine transform as indicated in equation (1.3.1) to obtain the overall spectral window for the computational process. This has been done (Simpson et al 1961b, Appendix K) for  $\omega_r = \pi/z$  which leads to an  $\lambda_t$  of  $\lambda_t$ : ..., 1, 0, -1, 0, 1, ...,  $\lambda_t$  and a correspondingly simple autocorrelation function. It can be seen that the Daniell estimate has parameters  $M$  and  $N$ , and therefore spectral windows were computed for several different  $\mathcal{M}$  and  $\mathcal{N}$  values. A few examples of the windows have been included in Figure 1.3.1 to 1.3.4 (Simpson et al, 1961b). These figures show that the windows are always non-negative, they tend to get squarer as the  $M/N$  ratio decreases and they are essentially non-oscillatory. The variance,  $\sigma_{\text{D}}^2$ , of the Daniell estimate has been worked out by E. A. Robinson (Simpson et al, 1961b, 1962a) and is

$$
\sigma_{\mathsf{D}}^2 = \frac{\pi}{2N\hbar^2} \int_{\omega_{\mathsf{D}} - \hbar}^{\omega_{\mathsf{a}} + \hbar} \Phi_{\tau}^2(\omega) d\omega
$$

where  $\mathbf{h}$  =  $\pi/\mathbf{M}$  and  $N$  is the number of data points. As an approximation to this we have used

$$
\sigma_{A}^{2} = \frac{\pi}{2Nh^{2}} \left[ \Phi_{D}^{2}(\omega) 2h \right]
$$

$$
\sigma_{A} = \sqrt{\frac{M}{N}} \Phi_{D}(\omega)
$$

Figure 1.3.5 shows a plot of the Daniell spectrum (solid line) of a typical noise record with dotted line denoting the approximate standard deviation,  $\mathbb{C}_{A}$ , plotted above and below the solid line. The spectra are plotted as histograms since the value at any one frequency is an estimate averaged over the spectral window width. We note that  $M$  is the number of spectral estimates between  $\omega$ <sup>-0</sup> and  $\pi$  . One can then see that the  $N/M$  ratio is an estimate of the number of cycles of a sine wave which the data affords and therefore an increase in  $N/M$  ratio (decrease in  $M/N$  ) means that one is looking at more cycles and can therefore make a better estimate of the frequency. This is, of course, just the uncertainty principle.

#### Spectrum and Benioff Response

It is important to remember that the data was recorded on a Benioff seismometer and that the spectrum we see is observed through the eye of the Benioff. The apparent spike at low frequency, .25 cps, is artificial since the Benioff cuts off the lows. The sharp cut off on the low frequency side of the major low frequency feature in the spectrum of Figure 1.3.5 and other spectra in Figures 1.3.6 to 1.3.9 is a result of the seismometer response and is not a real phenomenon. We notice from Figure 1.3.2 that there is essentially no energy at frequencies greater than 2.5 cps so that, with our sampline rate of 20 samples per second, there is no problem with aliasing of frequencies.







Figure 1.3 .2



**4.9**



Figure 1.3.4

 $\begin{array}{c} {\mathbb{C}} \\ 0 \end{array}$ 



POWER DENSITY SPECTRUM OF RECORD 1000

Cycles Per Second Times 10

Figure 1.3.5

pectrum of Record 1000 with standard deviation  $\tilde{p}$  lotted above and below the spectral estimate.



Figure 1.3.6 Power Density Spectra o<br>F Records 1000 to 1005

 $2\overline{3}$ 







 $\overline{V}S$ 



igure 1.3.9 Power Density Spectra of Records 204 and 233 (CHP 4 and CHP 31).<br>(Note: The spectra have different frequency scales.)

J ιJ.

## 1.4 Mathematical Generating Model for Microseisms

#### Stationary Time Series - Moving Summation and Decomposition

We have seen that microseismic noise can be considered at least as a wide sense stationary time series. With an additional assumption of an absolutely continuous spectral distribution (Doob, 1953) we can consider that the time series is generated by a moving average or moving summation which is written as a convolution. That is, the time series  $x_t$  can be generated by convolution of an uncorrelated or purely random series,  $\zeta_t$ , with a weighting function  $\forall i$ .

$$
\chi_t = \sum_{i=-\infty}^{\infty} w_i \zeta_{t-i}
$$

Since  $\zeta_t$  is at least uncorrelated and may be purely random, it is obvious that the autocorrelation of  $X_t$  will simply be the autocorrelation of  $w_i$ . Hence the spectral properties of  $X_t$  are defined by the wavelet  $\forall i$ . If the power density spectrum,  $\Phi(\omega)$  , of the time series or, equivalently, of  $\mathsf{W}_{i}$  can be factored

$$
\overline{\Phi}(\omega) = B(\omega) \overline{B(\omega)}
$$

and  $\mathcal{B}(\omega)$  has no poles or zeros in the lower half plane then

$$
B(\omega) = \sum_{\kappa_0}^{\infty} b_{\kappa} e^{i \omega \kappa}
$$

and

$$
W_{\kappa} = b_{\kappa}, \quad W_{\kappa} = 0 \text{ for } \kappa \omega
$$

(See Appendix E, Spectrum Factorization)  $b_{\kappa}$  is one sided and invertable and is called the minimum phase wavelet. The considerations

1. 
$$
\Phi(\omega) = 0
$$
 almost nowhere  
\n2.  $\int_{-\pi}^{\pi} I_{0} \Phi(\omega) d\omega > -\infty$   
\n3.  $\int_{-\pi}^{\pi} \Phi(\omega) d\omega < \infty$ 

must be met for  $\mathbf{b}_k$  to exist (Robinson, 1956). These conditions are discussed further in Appendix E.

If we assume that the above conditions are met for microseismic noise, we can choose a simple mathematical model for microseism generation. We can consider that microseisms can be produced by passing a train of white light (uncorrelated) impulses through a system whose transfer function is  $B(\omega)'$ . In block diagram form:



 $\mathbb{R}(\omega)$  corresponds to a realizable system since  $\forall x \in \mathbb{R}$  is a one sided wavelet. Spectrum factorization computations using the method of Kolmogorov as

described in Appendix **E** have been carried out on real microseismic noise. Figures **1.3.6** to **1.3.8** show the spectra and Figures 1.4.1 to 1.4.5 show some of the minimum phase wavelets and inverse minimum phase wavelets for several of the Logan and Blanca noise records.

## Autoregression, Probability Density and Edgeworth Series

Since the inverse minimum phase wavelet,  $\mathbf{Q}_{\kappa}$ , exists, we can represent the noise  $X_t$  as the autoregressive process

$$
\xi_t = \sum_{k=0}^{\infty} \alpha_k x_{t-k}
$$

where  $\xi_t$  is the white light series, and  $\alpha_k$  can be found from  $b_k$  by polynomial division (See POLYDV in Appendix **G).**

$$
A(\omega) = \sum_{k=0}^{\infty} \alpha_k e^{i\omega k} = B(\omega) = \frac{1}{\sum_{k=0}^{\infty} b_k e^{i\omega k}}
$$

Taking the Z transform,  $2 = e^{i\omega}$ 

$$
\sum_{k=0}^{\infty} a_{k} z^{k} = \frac{1}{\sum_{k=0}^{\infty} b_{k} z^{k}}
$$

Hence the white light series  $\zeta_t$  for the process can be found by convolution of  $\alpha_R$  with  $x_t$ . This computation has been done for most of the Logan and Blanca noise records and statistical tests have been made on the resulting white light series,  $\xi_t$ . The probability density of  $\xi_t$ for these records has been compared to the normal density using the steps

outlined in Appendix B. In most cases the comparison measure resulted in the probability of exceeding chi-squared being so small that it was very unlikely the density of  $\zeta_t$  was exactly normal. The numerical results summerized in Table 1.4.1 show that only four of the records pass the  $\chi^2$  test. The empirical densities, however, look so very nearly Gaussian (see Figures 1.4.6 to 1.4.12) that it seems likely that they can be expressed in terms of the Gaussian density with only small correction terms. (Note that we use the terms "Gaussian" and "normal" interchangeably throughout this section. Cramer (1951) gives the Edgeworth series expansion for the probability density  $f(x)$ 

$$
f(x) = C_0 \ \hat{f}(x) + \frac{C_1}{l!} \ \hat{f}'(x) + \frac{C_2}{2!} \ \hat{f}'(x) + \dots + \frac{C_n}{n!} \ \hat{f}'(x) + \dots
$$

where  $\hat{\mathcal{Y}}(x)$  is the Gaussian,  $\hat{\mathcal{Y}}(x) = \frac{1}{\sqrt{2\pi}} e^{-\frac{x^2}{2}}$ , and the superscripts denote differentiation. The  $C_n$  depend on the moments. The details of the applicability of the expansion and the computation of the moments and the C<sub>n</sub> appear in Appendix C. The first seven  $C^s$ ,  $C_o$  to  $C_o$  have been computed and the corresponding densities have been compared with the empirical density using the chi-squared measure of goodness of fit.

## Normality - Chi-Squared Test

Table 1.4.2 shows the results of the Chi-squared test of the comparison of the probability density of the white light series with the normal density and the higher approximations given by the Edgeworth series. The method of computation of the Chi squared value used here differs somewhat from the method mentioned in Appendix B. In Appendix B

we ignore the fact that the series undergoing the test is bounded and, after dividing up the normal density into **N** regions of equal area (probability), we count the number of data points which fall into each region. The approximation involving the terms in the Edgeworth series, including the normal approximations were compared directly to the empirical density, computed for **V** subregions over the interval in which the data fell. There was not attempt at division into regions of equal probability. For this case, where the chi squared value is computed directly from the probabilities, chi squared is

$$
\chi^{2} = \sum_{i=1}^{r} \frac{\left(P_{A_{i}} - P_{E_{i}}\right)^{2}}{P_{A_{i}}}
$$

where  $\mathbf{P}_{\mathbf{A}}$  is the probability that a value falls in the ith range using the approximation given by the Edgeworth series,  $P_{\epsilon_1}$ <sup>'</sup> is the empirical probability density for the same range,  $N$  is the number of data points which were used to compute the empirical density, and **Y** is the number of sub-regions used in forming the empirical density. There may be some bias in this method of computation if  $P_{A_i}$  and  $P_{E_i}$ are very small. For this reason the sub-regions are grouped together so that for every grouping the quantities  $P_{A_i} N$  and  $P_{E_i} N$  are both at least five. (This rule of thumb is given in Wadsworth and Bryan, **1961).** The grouping will reduce the number of degrees of freedom so that it becomes

$$
NDF = S-1-m
$$

where m is the highest moment used in the Edgeworth series and S is the total number of sub-groupings.  $\int$  is in general less than  $\mathsf{Y}$ . We note that this method compares the empirical density and the approximation about the normal density only over the region where the data actually exists and does not assume that the data is unbounded.

In computing  $P_{A_i}$  it was necessary to calculate at least five equally spaced points across the sub-region and integrate using Simpson's Rule. The estimate of the integral using just the center point was not accurate enough. (We note here that  $P_{\xi}$ <sup>'</sup> is a probability density and thus must be normalized such that its integral is equal to one.)

We see from Table 1.4.2 that, using the above method of comparison. most of the white light series are actually Gaussian (first approximation of Edgeworth series), and all can be fitted quite well using the third approximation or less. It is not disturbing that the fit gets poorer in some cases for higher approximations, since the series used is asymptotic and may oscillate.

Figures 1.4.6 to 1.4.12 show the empirical density as a solid line histogram and the Edgeworth approximation as a dotted line. The first approximation is the normal, the second approximation involves the third moment since  $C_0=1$ ,  $C_1=0$ , the third involves up to the fourth moment, etc. We can therefore say that the probability density of is,in most cases, Gaussian.

# Independence Tests

The  $\zeta_t$  are necessarily uncorrelated since the convolution of  $\chi_t$  with

has removed all the linear dependence. It is not necessary that the  $\zeta_t$ series be purely random or, equivalently, independent (unless the  $\xi_t$ are normally distributed, see section **2.3).** Independence tests are somewhat difficult because one has to show that the joint probability density for all  $\zeta_t$  factors in order to prove independence.

$$
P_{\xi_1,\xi_2,\dots,\xi_n}(x_1,x_2,\dots,x_n) = P_{\xi_1}(x_1) P_{\xi_2}(x_2) \dots P_{\xi_n}(x_n)
$$

Two tests for independence have been used on the  $\zeta_t$  from microseismic noise. The poker count test (Appendix **D)** is based on the fact that we can compute the a priori probabilities of occurrance of poker hands of various values from the assumption of independence of the series from which the hands are drawn. In this case the hands are assumed drawn from an infinite supply of integers with values **0** to **9** and hence the removal of a number does not change the probability of its occurrance. In the performance of the poker count test, the  $\zeta_t$  must be integers from **0** to **9** with equal probability, so the series with nearly Gaussian density must be mapped into a series with rectangular density. This mapping will not make the series dependent if it is independent and vice versa. Proof of this statement and the steps necessary for the poker count test are given in Appendix **D.** We may note that the poker count test is concerned with the joint density of up to five variables. The other test, the dependence measure related to the mean square contingence test, is also treated in Appendix **D.** It is simply a numerical measure of the factorization of the joint density of two random variables.

The measure, which we call the dependency, is zero is the variables are independent, and non-zero otherwise. Tests of numerical data are somewhat difficult since in almost no case will the dependency actually come out zero although it may be quite small. In order to see how small the dependency measure must be to indicate dependence, the test was run on the Rand random digits (Rand Corporation,  $1955$ ). These digits were generated by an independent process and are therefore suitable for testing purposes. A graph of the result of this test for different series lengths appears in Appendix D. For a length of 2500 the average dependency was about .0035. For dependent series such as the amplitude of the microseisms the dependency was about .25. The dependency value for the white light series, were between .0907 and .0039 and are tabulated along with the tests on the amplitudes in Table 1.4.1. Some output from the tests is shown in Figures 1.4.13 and 1.4.15. In some cases the dependency value was as low as that of the Rand digits and in others it was somewhat higher but not orders of magnitude higher. The figures mentioned above also show the results of the poker count test. In most cases a chi-squared comparison of the results is in the .1 or .05 acceptance region. The poker count test was also run on the Rand random digits. For these the chi-squared value was quite low and well within the accptance region.

## Mathematical Model

The independence tests performed on are certainly not exhaustive since the poker test treats up to fifth joint density and the mean square contingency treats only the second joint density. The results are

surprisingly good, however, particularly when we consider the error in the computation of the  $\zeta_t$  series introduced by the spectral estimation procedure, spectrum factorization, polynomial division and convolution. It is therefore claimed that the  $\left\{ \begin{matrix} 1 \\ 1 \end{matrix} \right\}$  series is essentially independent and the microseism generating model is now an independent white light series into a minimum phase system. '

A purely random series  $\zeta$  is ergodic and stationary. Further, the process of moving summation (convolution) is ergodic (Robinson, **1956, p. 116).** Ergodicity, for our purposes, means that the time averages and ensemble averages are equal with probability one (see also Section 1.2). Hence the estimation of the moments of the series **by** time averages for the expansion of the density in terms of the Gaussian is justified.

In summary, we have shown that microseismic noise can be considered stationary and ergodic with a nearly Gaussian probability distribution, The model for the generation is an independent white light series convolved with a minimum phase wavelet.



Independent White Microseismic Light Series - Nearly Microsett Noise Gaussian

$$
X_t = \sum_{k=0}^{\infty} b_k \xi_{t-k}
$$

## Generation of Artificial Microseisms

We are now in a position to generate microseismic noise artificially. The Rand random digits which are independent and equally likely were summed in groups of ten and the mean subtracted out to give, by the central limit theorem,zero mean normal variates. These variates are the Gaussian white light input to the minimum phase system. They are Gaussian because of the central limit theorm as mentioned above, and white because the independence of the variates guarentees that only the zero lag of the autocorrelation has a non-zero value and hence insures that all frequencies will be present in the same amount. The minimum phase system response, can be computed from real data by spectrum factorization (Appendix E). The artificial noise is then generated by convolution of the minimum phase wavelet with the Gaussian white light series. Figure 1.4.16 shows real and artificial microseismic noise with the same r.m.s amplitude plotted one above the other. It is difficult, if not impossible, to tell the difference between the two with the eye alone. The identification of the two traces has been deliberately omitted from the figure. The upper trace is actually the artificial noise. Since we have been able to show that microseismic noise can be decomposed into a white light series and a wavelet, and that the white light is fairly indpenedent and nearly Gaussian, our mathematical model is quite good, and thus our artificial microseisms are quite representative. In order to tell the difference between real and artificial microseisms we would have to decompose the series into a wavelet and white light and test the probability density against the normal density. If it is normal and not just "nearly" normal, the noise is

artificial. It is possible to overcome this difficulty **by** mapping the Gaussian series into a series with a probability density representative of the real noise, but this labor does not seem justified **by** the slight variation of the probability density from the Gaussian.

The chief use of the generating model is in the detection simulation studies in Chapter **3.** Several hours of consecutive noise are needed for these studies and only a few minutes of it is available from our records. Using the model discussed above we can generate the necessary amount of noise artificially and it will be typical of microseisms and nearly indistinguishable from them.

It is also possible to generate three component artificial noise. The bind here would appear to be in simulating the coherency between the various components. However it has been shown (Simpson et al, **1962)** that one can generate pairs of white light series with controlled coherency at zero phase. **A** simple extension of this to three series with controlled coherencies is given in Appendix F. One can therefore specify the coherencies between pairs of the three series, generate three white light series with these coherencies, and convolve each of the series with a different wavelet to obtain three component simulated coherent microseismic noise.

#### TABLE 1.4.1

## SUMMARY OF RESULTS OF NORMALITY AND DEPENDENCY TESTS ON AMPLITUDE SERIES AND WHITE LIGHT SERIES.



PROBABILITY OF EXCEEDING CHI SQUARE LISTED AS .0000 IS ACTUALLY LESS THAN .000032, BUT NOT ZERO.

#### TABLE 1.4.2

#### EDGEWORTH SERIES RESULTS



DEGREES REFERS TO THE NUMBER OF DEGREES OF FREEDOM FOR THE LOWEST APPROXIMATION NUMBER FOR WHICH THE PROB ABILITY OF EXCEEDING CHI-SQUARED IS GREATER THAN .01.

8

 $\mathcal{L}^{\text{max}}_{\text{max}}$  and  $\mathcal{L}^{\text{max}}_{\text{max}}$ 



Figure **1.4.1**



Figure **1.4.2**



Figure 1.4.3



Figure **1.4.4**

 $\mathcal{L}$


Figure 1.4.5

 $\mathbb{C}$ 



Figure 1.4.6 Empirical Probability Density of White Light Series<br>of Record 1000 With First Five Edgeworth Series Approximations.

 $\overline{\phantom{a}}$  $\tilde{=}$ 



Figure 1.4.7 Empirical Probability Density of White Light Series **of** Record 1001 With First Five Bdgeworth Series Approximations.

 $\sim$   $\!\!1$  $\zeta$ 



Figure 1.4.8 Empirical Probability Density of White Light Series of Record 1006 With First Five Bdgeworth Series Approximations.

 $\overline{\phantom{a}}$  $\mathbb{C}^n$ 



NGE 157<br>ANGE 157 (2000 HEAR) UNIT VARIANCE INTS BY 10<sup>2</sup> Figure **1.4.9** Empirical Probability Density of White Light Series of Record 1007 With First Five Bdgeworth Series Approximations.

 $\overline{\phantom{a}}$  $\sim$ 



Figure 1.4.10 Empirical Probability Density of White Light Series<br>Of Record 1008 With First Five Edgeworth Series Approximations.

 $\frac{1}{\infty}$ 



Figure 1.4.11 **Empirical Probability Density of White Light Series**<br>of Record 1026 With First Five Edgeworth Series Approximations.

 $\sim$  $\circ$ 



Figure 1.4.12 Bmpirical Probability Density of White Light Series of Record 1027 With First Five Edgeworth Series Approximations.

 $\frac{8}{2}$ 

# Figure 1.4.13

**ANALYSIS** OF WHITE LIGHT SERIES **OBTAINED** BY **CONVOLVING** THE INVERSE OF **THE** MINIMUM **PHASE** WAVELET OF RECORD **1000** WITH THE **ORIGINAL** RECORD

COMPARISON OF ACTUAL DISTRIBUTION AND NORMAL DTSTRIBUTION

NUMBER OF RANGES= **51 LENGTH** OF SERIES= **2702** DEGREES OF FREEDOM= 48 **MEAN OF SERIES= -0.10384890E 03** STANDARD DEVIATION= 0.75864953E 05

HIGHER CENTRAL **MOMENTS** THIRD **MOMENT= 0.91304071E** 14 FOURTH **MOMENT= 0.17391028E** 21 FIFTH **MOMENT=-** -010809396E **25** SIXTH MOMENT= **0.17594533E 32**

**EXPECTED COUNT=** 52.9804

CHI-SQUARE= 0.11462693E 03 PROBABILITY OF EXCEEDING CHI-SQUARE IS LESS THAN 0,00032

### POKER **COUNT** TEST **RESULTS**



MEAN **SOUARE** CONTINGENCY= **0\*88167071E-01**

**DEPENDENCY MEASURE=** 0.97963411E-02

### PROBABILITY DISTRIBUTION

NUMBER OF VALUES IN EACH OF 100 EQUALLY SPACED **RANGES** FROM -0.53663570E 06 TO 0.43644589E **06. 2702 VALUES IN** ALL.



Pigure 1.4.14

ANALYSIS OF WHITE LIGHT **SERIES OBTAINED** BY **CONVOLVING** THE **INVERSE** OF THE MINIMUM **PHASE** WAVELET OF RECORD **1006** WITH THE **ORIGINAL** RECORD

COMPARISON OF **ACTUAL DISTRIBUTION AND** NORMAL **DISTRIBUTION**

 $51$ NUMBER OF **RANGES=** LENGTH OF SERIES= 2682 **DEGREES** OF FREEDOM= 48 **MEAN** OF **SERIES= 0,17902389E 03 STANDARD DEVIATION=** 0o71888679E **05**

HIGHER **CENTRAL MOMENTS** THIRD **MOMENT= -0O47103929E** 14 FOURTH **MOMENT= 0.22192675E** 21 FIFTH **MOMENT= -0o62127688E 26** SIXTH **MOMENT= 0O67908355E 32**

**EXPECTED COUNTs 52.5882**

**CHI-SQUARE= 0.61046970E** 02 PROBABILITY OF **EXCEEDING CHI-SQUARE<sup>=</sup> 0.96320E-01**

 $\sim$  .

# POKER **COUNT TEST RESULTS**



MEAN SQUARE CONTINGENCY= 0.73803157E-01

**DEPENDENCY MEASURE= 0.82003506E-02**

 $\sim 100$ 

### PROBABILITY **DISTRIBUTION**

**NUMBER** OF **VALUES IN EACH** OF **100 EQUALLY SPACED RANGES** FROM **-0o73412665E 06** TO 0o48402021E **06. 2682 VALUES IN ALL\***



الموارد سالمصابك بالمعارض

Figure 1.4.15

**ANALYSIS** OF WHITE LIGHT **SERIES OBTAINED** BY **CONVOLVING** THE **INVERSE** OF THE **MINIMUM PHASE** WAVELET OF RECORD **1026** WITH THE **ORIGINAL** RECORD

COMPARISON OF **ACTUAL DISTRIBUTION AND** NORMAL **DISTRIBUTION**

**NUMBER** OF **RANGES=** 55 LENGTH OF SERIES= 3082 **DEGREES** OF FREEDOM= **52** MEAN OF SERIES= **0.29668643E** 02 S 049980906E **05 STANDARD DEVIATION=**

HIGHER **CENTRAL MOMENTS** THIRD **MOMENT= 0.36927477E** 14 FOURTH MOMENT= 0.41691343E 20 FIFTH **MOMENT= 0.39579482E** 25 SIXTH **MOMENT=** 0,22342489E **31**

**EXPECTED.COUNT= 56.0364**

**CHI-SQUARE= 0.15871704E 03** PROBABILITY OF **EXCEEDING CHI-SQUARE** IS **LESS THAN 0.00032**

### POKER **COUNT TEST RESULTS**



### **MEAN SQUARE CONTINGENCY= 0&43508112E-01**

**DEPENDENCY MEASURE= 0** .48342347E-02

#### PROBABILITY **DISTRIBUTION**

**NUMBER** OF **VALUES IN EACH** OF **100 EQUALLY SPACED RANGES** FROM **-0.27321346E 06** TO **0.35513622E 06. 3082 VALUES IN ALL\***









# **1.5** Cross-Series Properties

The availability of simultaneous three component seismic noise records from different stations affords opportunity for cross correlation and cross-spectral analyses. Techniques similar to those of autospectral analysis have been worked out and programmed for high speed digital computers. The major computational difference is the need for a sine transform in addition to the cosine transform since the cross correlation is not in general an even function. Knowing the sine and cosine transforms of the cross correlation it is easy to compute the magnitude cross power and phase spectra, and it is also useful to compute the coherency. The development of the usual expression for coherency can be done quickly for transients and then carried over to discrete time for our case.

# Cross Correlation, Cross Power and Coherency

For two transients  $\chi(t)$  and  $\chi(t)$  the cross correlation is

$$
\varphi_{xy}(\tau) = \int_{-\infty}^{\infty} \chi(t) y(t+\tau) dt
$$

The cross power spectrum is then the Fourier transform

$$
\oint_{xy} (\omega) = \frac{1}{2\pi} \int_{-\infty}^{\infty} \int_{x}^{\infty} r(x) e^{i\omega \tau} dx = \frac{1}{2\pi} \int_{-\infty}^{\infty} x(t) y(t-\tau) e^{i\omega \tau} d\tau d\tau
$$

with the change of variables  $r = t + \gamma$  this becomes

$$
\Phi_{xy}(\omega) = \frac{1}{2\pi} \int_{-\infty}^{\infty} \chi(t) e^{-i\omega t} dt \int_{-\infty}^{\infty} y(r) e^{i\omega r} dr
$$

 $hence$ 

$$
\Phi_{x,y}(\omega) = 2\pi \overline{F_x(\omega)} F_y(\omega) \qquad (1.5.1)
$$

where  $F_{\chi}(\omega)$  is the Fourier transform of  $\chi(\star)$ ,  $F_{\chi}(\omega)$  the Fourier transform of  $y(t)$ , and the bar denotes complex conjugation. The auto-power spectra are found to be, by similar treatment,

$$
\Phi_{xx}(\omega) = 2\pi F_{x}(\omega) F_{x}(\omega)
$$
  

$$
\Phi_{yy}(\omega) = 2\pi F_{y}(\omega) F_{y}(\omega)
$$

The coherency is then usually defined as

$$
C_{oh_{xy}(\omega)} = \frac{\sqrt{\Phi_{xy}(\omega)}|}{\sqrt{\Phi_{xx}(\omega) \Phi_{yy}(\omega)}}
$$

$$
= \frac{|\overline{F_{x}(\omega)} F_{y(\omega)}|}{\sqrt{\overline{F_{x}(\omega)} \overline{F_{y}(\omega)} \overline{F_{y}(\omega)} \overline{F_{y}(\omega)}} = 1
$$

This definition is not particularly useful since  $\text{Coh}_{\mathbf{X}\mathbf{y}}(\omega)$  is always

one. If the cross-correlation is weighted by some function, such as the Daniell weighting function (Section 1.3), the coherency is not necessarily one and has some meaning as a measure.

We define the normalized cross power vector  $N(\omega)$ 

$$
N(\omega) = \frac{\Phi_{xy}(\omega)}{\sqrt{\Phi_{xx}(\omega)} \Phi_{yy}(\omega)}
$$

where  $\Phi'_{xy}(\omega)$  now takes into consideration the weighting function  $W(\tau)$ .

$$
\Phi'_{xy}(\omega) = \frac{1}{2\pi} \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} x(t) y(t+\tau) dt e^{i\omega \tau} W(\tau) d\tau
$$

$$
\Phi_{xy}(\omega) = F_x(\omega) F_y(\omega) * w(\omega)
$$

where  $W(w)$  is the Fourier transform of  $W(\tau)$  and the asterisk denotes convolution.  $\overline{\varphi}_{xy}(\omega)$  is in general complex, hence  $N(\omega)$  is truly a vector. The coherency is then defined

$$
\mathsf{coh}_{\mathsf{x}\mathsf{y}}(\omega) = |N(\omega)|
$$

# Daniell Window and *M/N* Ratio

The treatment is almost identical for discrete time. The complete transient cross correlation for the two series  $\lambda_t$  and  $\mu_t$  each of points is

$$
\psi_{xy}(\tau) = \frac{1}{2N-1} \sum_{t=-(N-1)\tau}^{N-1\tau} \chi_t \, y_{t+\tau} \quad , \quad \tau = 0, \pm 1, ..., \pm (N-1)
$$

and the cross power spectrum with the Daniell weighting function is

$$
\Phi'_{xy}(\omega) = \frac{1}{2\pi} \sum_{T=-\{N-1\}}^{\{N-1\}} \varphi_{(T)} \left( \frac{\sin \frac{\pi \tau}{M}}{\frac{\pi \tau}{M}} \right) e^{i \omega \tau}
$$

We shall take  $W = n \omega_0$  with  $\omega_0 = \pi/m$  where M is the Daniell parameter, and  $h = 0, 1, 2, \ldots$ , M . We have seen in Section 1.3 that, for  $N/M$ large, the Daniell window is nearly rectangular. With  $W_0$ :  $\pi / M$  the windows for neighboring spectral estimates  $K$   $W_o$  and  $(K+1)$   $W_o$ overlap by about 50%. The Daniell window averages the sine and cosine transforms over the window width and consequently averages the cross power vector,  $N(\omega)$  . We see, therefore, that  $|N(\omega)|$ , the coherency, is less than or equal to one. If the  $N(\omega)$  vector changes direction rapidly over the band  $\omega \pm \frac{\pi}{\lambda}$  the vector averaging will tend to cancel out and the coherency will be low, and if the vector direction is not changing or changing only slightly, the coherency will be high. Thus the coherency as we use it is a measure of how rapidly the cross power phase is changing. If the records being cross correlated are identical, the phase spectrum is zero and the coherency is one. (Actually the coherency may be slightly less than one since the Daniell window is not quite

rectangular.) If the records are different, the coherencies will be low unless there are some bands of frequencies where the phase remains relatively constant.

# Cross Spectra **of** Different Components at the Same Station

Figures **1.5.1** to **1.5.3** show the results of the cross spectral computations between different components at the same station. The graphs in the figures are identified individually with the two record numbers of the data used, the indices of the first and last points of the data for each record and the Daniell parameter,  $M$ . In most cases, no computation has been done for frequencies above five cps. The recordings at any one station were made within a fraction of a wavelength of any wave of interest so that no compensation need be made for linear phase shifts due to spatial separation.

Figure **1.3.1** shows the cross-spectra of the components of the noise recorded before the Logan shot **1902** km from the shot (records **1000,** 1002 and 1004). The only really prominent feature of this set of computations is the low frequency spike which is the tail end of the well-known oceanic microseisms. The Benioff instrument cuts off fairly sharply at low frequencies so that this spike is somewhat artificial in that its low frequency side is simply instrument cutoff, but that sharpness of the higher frequency side must be a real phenomenon. The phase spectrum does not show the expected **900** phase shift for Rayleigh waves, but this may be explained **by** the fact that the instrument characteristics are changing rapidly here and are hence possibly non-uniform from instrument to instrument. None of the frequencies with fairly high coherency seem to

have phases corresponding to any known wave type. We note that the phases have been plotted to fall between  $+ \pi$  and  $- \pi$ .

Figure **1.5.2** shows the cross-spectra of the components of the noise before the Logan shot 2111 km from the shot point (Records **1006, 1008** and **1010).** The **1008-1010** set of graphs have high coherence and power at 1.9 cps, but the phase is  $-\pi$  which does not pin down any wave type. The peak at 2.1 cps has a phase closer to **-900** which could conceivably be a Rayleigh wave. The **1006-1010** set of graphs has reasonably coherent peaks at **.6,** 1.4 and **1.9** cps. The **.6** and 1.4 cps peaks are nearly in phase and could, therefore, be Love waves. The **1.9** cps peak is another of the many bands which are fairly coherent but have phase relationships which are not indicative **of** any particular wave type.

Figure **1.5.3** shows the cross spectra of the noise recorded before the Blanca shot **1610** km fyom the shot (records **1026, 1028** and **1030).** There are possible Rayleigh waves at **1** and 2 cycles per second, but the coherencies are somewhat low.

Figure 1.5.4 shows the auto spectra of the records used in the cross spectral computations. They are included for convenient reference.

It seems that, in view of the above results, the model of a single band of surface waves from one direction is entirely too simple. It is much more likely that there are many surface waves of several frequencies coming from several sources. For a few stations quite close to the coast it may be possible to complicate the model to take care of surface waves from a few directions, and produce some believable results. However, the stations for which we have good noise data are very far inland, nearly equi-distant from the Atlantic and Pacific coasts. Thus, sources from the

Atlantic, Pacific, Gulf and Great Lakes may produce microseisms which will be recorded with nearly the same amplitude at these inland stations. On top of this we have local sources which confuse the issue considerably. The higher frequency bands at 1.4 and 2.0 cps were seen in the last two sections to have no particular directional properties and to have no simple amplitude dependence on distance from water wave sources. We conclude that there are of local origin and may be isotropic. Even a fairly complicated model taking into account many sources may not fit the data too well, and would certainly require a lot of labor to use.

# Cross Spectra of Like Components at Different Stations **-** Linear Phase Shifts

The coherency measure used causes some difficulty if the two series are shifted in time, since a time shift will result in a linear phase shift. For example,  $e^{i\omega t}$  has zero phase at time  $t=0$  but at a later time the phase is  $\omega^{\dagger}$ . If the time shift is large, the phase changes over the small band of frequencies  $W^{\pm}$ <sup> $\pi/\mathcal{M}$ </sup> will be large and will tend to reduce the coherency estimate. If meaningful coherency values are to be obtained one must line up the records properly in time before computing the cross correlation. This procedure assumes that the relative time shift is known and this is not always the case. For three component records at one station there is no difficulty since a line up in absolute time is all that is necessary. However, if one is trying to follow a wave packet across considerable distance **by** cross correlation and coherency measures, difficulties arise. If the records are lined up in absolute time, the relative time of the maximum of the cross correlation may give an idea of the arrival time differences, but the coherency will not

necessarily be large in the range of the frequencies which comprise the wave packet. If the records are shifted the amount,  $\tau$ , indicated by the maximum of the cross correlation and then cross correlated, the coherency in the frequency region which caused the maximum will certainly become larger, but there may have been features in the original record other than the wave packet which caused the maximum. Hence we have still not identified the wave packet or its relative time shift. The magnitude of the time shift for any particular wave packet will of course depend on the velocity,  $V_i$  of the packet, on the distance between the stations, X , and on the direction of travel of the wave relative to a line between the stations. The time shift can therefore vary from  $t=0$ , if the waves are travelling perpendicular to the line between the stations, to  $t = \frac{x}{v}$ , if the waves are parallel to the line. The problem is complicated by the existence of many waves of different frequency of waves of the same frequency travelling in different directions. In even the simple case of a single wave packet dispersion may disrupt the coherence.

There is another scheme to find the appropriate time shifts which is a bit more promising than the cross correlation method. If the cross correlation is computed and not weighted by the Daniell factor, the sine and cosine transforms will not average the cross power vector over the Daniell window width. The cross power vectors can then be rotated by phase shifts corresponding to known time shifts in the frequency range of interest and averaged in this range. This is done for several time shifts and one looks for the time shift corresponding to the largest resultant of the averaged vectors. This should be close to the shift

necessary to maximize the coherency in the band of frequencies when the Daniell window is used.

Some time shifting experiments have been done using data from two different stations. Cross correlation and cross spectral computations have been carried out on like components at different stations using the methods described above. Figure **1.5.5** shows the complete cross correlation of records **1000,** the noise before the Logan shot **1902** km -from the shot point, and record **1006,** the noise before the Logan shot 2111 km from the shot point. The two records were lined up in absolute time before the computation. If most of the energy was travelling in one direction we would expect the cross correlation to have a pronounced maximum, but not necessarily for zero lag. There is no such maximum in Figure **1.5.5.** (The correlation is the transient cross correlation and so dies off to zero at the ends.) If the energy were coming directly from one station to the other at about **3** km/sec it would take about **70** seconds or 1400 data points. The correlation covers from minus to plus **2999** lags and should show a maximum if one were present. It is, of course, possible that a maximum occurs for one frequency and that it is masked **by** the presence **of** other frequencies. To check this for the more energetic bands, the data was band pass filtered before correlation. Figures **1.5.6** and **1.5.7** show the cross correlation for pass bands centered at 1.4 cps and 2.0 cps. The results are perhaps a bit disappointing but not totally unexpected. The cross correlation for the 1.4 cps band is exceedingly sinusoidal. This can, of course, happen if the band is too narrow, but we expect something more like the figure for the 2 cps pass band which

shows a beating between the frequencies present. It is not possible to pick a maximum on either of these figures with any certainty. If the energy is contained in such a narrow band as the 1.4 cps correlation indicates, the signal is not random enough for coherency to have any meaning.

Some time shifting was also done to maximize the coherency by looking for a linear trend in the phase. Figures 1.5.8 and 1.5.9 show cross spectral results for records 1000 and 1006 for several different time shifts. The frequencies about 1.4 and 2.0 cps were checked for a linear trend and appropriate shift made. The coherency was increased at these frequencies for the time shift indicated. The shifts were +1.5 seconds (that is, record 1000 has been shifted such that its absolute time origin,  $\top$ , lines up with absolute time  $T + 1.5$  seconds on record 1006) and  $-2.5$  seconds. In view of the cross correlation results, it does not seem that these time shifts, even though they increase the coherency, have any particular physical interpretation in terms of velocity and direction of travel of particular waves. If the 1.4 and 2.0 cps are from local sources (and there must be many of these local sources across the country to explain the occurrance of the spectral lines at different stations) we would not expect the time shifts to have any significance since the lines are narrow and the sources isotropic. With such narrow band signals we can expect the coherency to be high for shifts which are integer multiples of the wave period. We can see that time shifting experiments are not particularly fruitful for the narrow band signals or for the bands when the instrument characteristics change so rapidly with frequency that a mismatch between instruments is probable. The experiments are more suitable for long period records where local sources play a smaller part.

Some cross spectral computations were also done on some data from the WMSO linear array. Simultaneous sections of noise were used with no time shifting. The noise from the first instrument in the array was cross correlated with the noise from several other instruments in the array. The results are shown in Figures **1.5.10** to **1.5.15.** Again we see that at the frequencies with high coherence the phase is not changing rapidly. Figures **1.5.10** and **1.5.11** have a Daniell parameter of 400 and a slightly different frequency scale from Figures **1.5.13** and 1.5.14 which have a Daniell parameter of 200. The smaller Daniell parameter will take averages over wider bands and the resulting coherencies and phases will not be quite as jagged as those for a Daniell parameter of 400. Auto spectra are shown in Figures **1.5.12** and **1.5.15.** When the coherency is high, we tend to say that the waves at that frequency are travelling at right angles to the array and there is no linear phase shift to disrupt the coherency computation. The phase spectra also show in some cases linear trends over bands of frequencies which are of course accompanied by low coherencies. A time shift would bring up the coherency and indicate the direction of travel of the source waves for these bands.

A much more sophisticated analysis of array data is needed before any reliable results can be stated. Simulation studies of the sort described in Chapter **3** would be of interest with the array recordings time shifted (delayed) to minimize the noise and thus utilize the directional properties of the array. Similar studies could also be done with data from a two dimensional array.



Figure 1.5.1 Cross Spectra of Different Components at the Same Station



Figure **1.5.2** Cross Spectra of Different Components at the Same Station



Figure **1.5.3** Cross Spectra of Different Components at the Same Station



Figure *1.5.4* Auto Spectra

 $\mathbb{G}$ 



Figure 1.5.5 Complete Transient Cross Correlation of Records 1000 and 1006



Figure **1.5.6** Complete Transient Cross Correlation of Records 1000 and 1006 Band Pass Filtered at 1.4 Cycles Per Second



Figure *1.5.7* Complete Transient Cross Correlation of Records 1000 and 1006 Band Pass Filtered at **2.0** Cycles Per Second



Figure *1.5.8* Cross Spectra of Records 1000 and 1006 For Indicated Time Shifts



Figure 1.5.9 Cross Spectra of Records 1000 and 1006 For Indicated Time Shifts







Figure 1.5.11 Cross Spectra of Array Elements



2.0T



Figure 1.5.13 Cross Spectra of Array Elements






## 2. PREDICTION OF **MICROSEISMS**

2.1 Prediction **by** Minimization of Mean Squared Error

# Prediction and the First Motion Interval

Elementary considerations of the possible differences between the signals from earthquakes and the signals from underground explosions were based on the obvious differences in the source mechanisms. An explosion should give an initial compression whereas an earthquake, being a shearing source, should give compressions or rarefactions depending on the position of the observer relative to the fault plane and the direction of slip along the plane. A group of recording stations around a source should therefore all record initial compressive first motion for an explosion, but would vary if an earthquake were the source. Granting the first motion criterian is legitimate, there is still the problem of identifying the first motion on the record when the signal is corrupted by noise. The problem is somewhat simplified by the fact that, even though its pulse may be small, the first motion is followed by stronger P waves which are easily discernible in the noise. These P waves therefore allow us to say approximately where in time the first motion pulse arrived. If we could by some means predict what the noise would be in a small interval preceeding the strong P waves and subtracted the predicted noise from the signal plus noise, we would be left with the uncorrupted signal and could make definite statements concerning the direction of first motion. Figure 2.1.1 illustrates this idea with the assumption of perfect prediction of the noise.

In general, of course, we cannot predict perfectly, but a good prediction could possibly increase the signal to noise ratio to a point where there would no difficulty in picking out the first motion direction. We will therefore wish to express the predictability of the noise in terms of signal to noise ratio imprevement. Evaluation of the effectiveness of the scheme can be done by prediction studies of the noise alone without reference to any particular signal. The only parameter we need is time length over which we must predict. This will be called the prediction distance and it will be denoted by  $K$  in the following analysis.

We wish to form a linear operator which will predict the "future" of a record,  $X_i$ , from its "past" and possibly from the past of other related records (e.g. three components at one station). We note that even though we are not necessarily operating in real time it is necessary that we use only the past as a basis for prediction since the past is noise alone and the future is signal plus noise. We shall present the analysis for the formation of a linear operator operating on three records to predict one of the three. The expressions will reduce simply to the case of self prediction, the prediction of one record from itself. The analysis has been done (Wadsworth et al, 1953) for the two dimensional case and the simple extension to three dimensions is given here.

The requirement that the record  $X_i$  be predicted from itself and from  $\forall i$  and  $\sum_i$  can be stated by the regression function (Wadsworth et al, 1953). *M* A.

$$
\hat{x}_{i+k} = d + \sum_{s=0}^{m} \alpha_s x_{i-s} + \sum_{s=0}^{m} b_s y_{i-s} + \sum_{s=0}^{m} c_s z_{i-s}
$$

where  $\hat{\chi}_{i}$ + $\kappa$  is the predicted value of the  $X_i$  time series K time units ahead. One time unit is simply the sampling period and is .05 seconds for the Logan and Blanca records. The  $X_i$  are the actual noise  $d$ ,  $a_5$ ,  $b_5$  and  $c_5$  constitute the linear operator which values and must be determined. The criterion used in this determination is the Wiener mean squared error criterion where we wish to minimize the sum of the mean squared error between the actual and predicted  $X_i$  series. This means. of course, that we have to know what the future is of the noise above. Hence a long series of pure noise is arbitrarily divided into past and future and the operator formed. The operator, under the assumption of stationarity of the time series, can then be used on the portion of the noise preceding the first motion to predict the noise in the first motion interval.

### Mean Squared Error Techniques for Three-Dimensional Case

The sum of the squared error is taken over the operator interval length from  $i + K = N$  to  $i + K = N + n - 1$ a duration of n time units. Thus we minimize  $\Gamma$  where

$$
I = \sum_{i=N-K}^{N+n-1-K} (X_{i+k} - \widehat{X}_{i+k})^2
$$

$$
T = \sum_{i=N-k}^{N+n-1-k} \left[ x_{i+k} - (d + \sum_{s=0}^{M} a_s x_{i-s} + \sum_{s=0}^{m} b_s y_{i-s} + \sum_{s=0}^{M} c_s z_{i-s} \right]^{2}
$$

with respect to  $d$ ,  $a_5$ ,  $b_5$  and  $c_5$ . This is done by setting the partial derivatives with respect to  $d$ ,  $a_s$ , b<sub>s</sub> and  $c_s$  equal to zero for all S. The resulting set of 3M+4 equations for the **3** M+4 operation coefficients is

$$
nd + \sum_{s} \left[ a_{s} \sum_{i} x_{i-s} + b_{s} \sum_{i} y_{i-s} + C_{s} \sum_{i} z_{i-s} \right] = \sum_{i} x_{i+s}
$$
\n
$$
d \sum_{i} x_{i-r} + \sum_{s} \left[ a_{s} \sum_{i} x_{i-s} x_{i-r} + b_{s} \sum_{i} y_{i-s} x_{i-r} + C_{s} \sum_{i} z_{i-s} x_{i-r} \right] = \sum_{i} x_{i-r} x_{i+s}
$$
\n
$$
d \sum_{i} y_{i-r} + \sum_{s} \left[ a_{s} \sum_{i} x_{i-s} y_{i-r} + b_{s} \sum_{i} y_{i-s} y_{i-r} + C_{s} \sum_{i} z_{i-s} y_{i-r} \right] = \sum_{i} y_{i-r} x_{i+r}
$$
\n
$$
d \sum_{i} z_{i-r} + \sum_{s} \left[ a_{s} \sum_{i} x_{i-s} z_{i-r} + b_{s} \sum_{i} y_{i-s} z_{i-r} + C_{s} \sum_{i} z_{i-s} z_{i-r} \right] = \sum_{i} z_{i-r} x_{i+r}
$$

*or r=o to* **<sup>M</sup>** where summations over  $i$  are from  $i = N - \kappa$  to  $i = N + n - 1 - \kappa$ , and summations over S are from S **=o** to **S=** /: . We write this as the matrix equation

$$
RA = B \tag{2.1.1}
$$

where  $R$  is a  $3M+4$  by  $3M+4$  symmetric correlation matrix, each element depending essentially on different lags of the auto and cross correlations of  $X_i$ ,  $Y_i$  and  $\overline{z}_i$ . A is the  $3m+4$  by L solution matrix where each column of  $A$  is the prediction operator  $(\alpha_0^k, ..., \alpha_k^k)$  $b_{\delta}^{K}$ ,  $b_{\delta}^{K}$ ,  $c_{\delta}^{K}$ ,  $\ldots$ ,  $c_{\delta}^{K}$ ,  $d^{K}$  for different prediction distance K, and K takes on  $L$  different values.  $A$  is obtained by inversion of the  $\overline{R}$  matrix.

$$
A = R^{-1} \beta
$$

B is an L by  $3M+4$  matrix, where each column of  $\overline{3}$  is the right hand side of the equation for a different  $\kappa$ . The matrix equation can be partitioned as shown below



If we donote the auto correlation or Toeplitz matrix by



where  $r_i$  is the auto correlation for the  $j$ <sup>+</sup> $h$  lag we see that the diagonal submatrices of  $\mathbb{R}$  in equation (2.1.1) are not quite auto correlation matrices because the terms along diagonals of the submatrices are summed over different intervals. If the operator interval length,  $\hbar$ , is large, the diagonal submatrices are only very slightly different from auto correlation matrices and approach this as  $n \rightarrow \infty$ If we take the one dimensional zero mean case ( $b_5 = C_5 = d \cdot O$ ) with **h** large, the problem becomes the same as that treated by Levinson (1949).

# Predictability and the Percent Reduction

A measure of how well the prediction operator performs its task is the percent reduction,  $R_{\rho}$  . This quantity is defined (Wadsworth et al, 1953) as

$$
R_p = 100 \left( 1 - \frac{I_m}{I_o} \right)
$$

where  $\mathbf{I}_{\mathbf{m}}$  is the value for  $\mathbf{I}$  for the operator used and  $\mathbf{I}_{o}$  is a measure of the sample variance over the same interval.

$$
I_o = \sum_i (\mathsf{X}_{i+\kappa} - \overline{\mathsf{X}})^2
$$

If we think of  $I_0-I_M$  as a measure of the variance of the prediction we can see that the percent reduction is a measure of the amount of power which can be predicted. In terms of the signal to noise ratio, if we take  $S$  as a general signal and  $N$  the noise, then before filtering we have

$$
\left(\frac{S}{N}\right)_{\text{Berges}} = \frac{S}{\sqrt{\frac{1}{n} \mathcal{I}_o}}
$$

and after filtering

$$
\left(\frac{S}{N}\right)_{\text{ATTER}} = \frac{S}{\sqrt{\frac{1}{n} \mathbf{I}_{m}}}
$$

Hence

$$
\left(\frac{S}{N}\right)_{A \text{ATE}} = \frac{1}{\sqrt{1 - \frac{R_P}{100}}} \left(\frac{S}{N}\right)_{B \text{ FORF}}
$$

### Prediction Computations

In order to test the predictability, then, one must take a section of noise record, divide it into past and future and form the  $\kappa$  and  $\beta$ matrices given in equation (2.1.1). The  $\mathcal{R}$  matrix is inverted and  $\mathcal{R}$ <sup>\*</sup> is multiplied by  $\beta$  . The columns of the resulting  $A$  matrix are the operators or filters for different prediction distance  $K$ .  $h$  predictions for a given K are made **by** moving the operator along the real data for successive points. The prediction error,  $\mathbf{I}_m$  for this  $\kappa$  can then be

formed and, with  $I_0$  for the same  $h$  points, the percent reduction can be computed. This is done for each operator so that the percent reduction as a function of  $K$  can be obtained.

This procedure has been programmed for the IBM 709-7090 computers. Computation has been done for one dimension with several  $M$  values with  $K = i$  **to 30** and for three dimensions with  $M = 30$  also for  $K = 1$  to 30. The results of the one dimensional experiments are shown in Figures 2.1.2 to 2.1.4. The percent reduction should increase with increasing length of operator (M value) and does in all cases computed. For an infinite length operator the percent reduction must decrease monotonely with  $\kappa$ (Robinson 1954, p. 148) which does not occur in the cases shown. This is obviously due to the short operator lengths used in the computations, and we can be sure that higher percent reduction would be obtained with longer operators. The spectra of the records (Figures 1.3.6 to 1.3.9) show that most of the energy is crowded into a few narrow bands, the lowest frequency being about 1 cps. It would be best to have operator lengths covering a few wave lengths of the major frequency components which in this case would be about three seconds or at least 60 terms. The method of solution for the operators then involves inversion of a 60 by 60 matrix which starts to suffer from round off error.

We note that in all cases the percent reduction falls off rapidly at first and then has one or more plateaus. The Cherry Hill Park records remain fairly predictable out to three seconds, maintaining a percent reduction of about 50. This is attributed to the narrowness of prominent spectral lines of these records. (A spike in the frequency domain represents

a sine wave and can be predicted exactly with a two term operator.)

If a typical wave length of the first motion is established at 1 second the corresponding prediction distance for the C.H.P. records would be 10 units. This would give a signal to noise ratio improvements of 1.4 and 1.3 for C.H.P. 31 (record 237) and C.H.P. 4 (record 204) which is not significant.

The Logan 1902 records show a plateau effect in the percent reductions but the initial fall is more pronounced than in the C.H.P. records. The vertical is the most predictable component and a 20 term operator gives a signal to noise improvement of only about 1.3 for 1 second (20 units).

We have seen that the predictability in the one dimension or self prediction case is not particularly significant. However, one might expect that the use of information from more than one component would do somewhat better if the components used are related. The analysis for three components has been shown and was programmed for the IBM 709-7090 computers.

The precent reduction for  $M$  values of 5, 10, 15 and 20 (corresponding to operator lengths of 16, 31, 46 and 61) for the prediction of the vertical component, Logan 1902 **km,** record 1002 from itself and the two horizontals is shown in Figure 2.1.5. Comparison of this figure with Figure 2.1.3, the self prediction results, shows an almost imperceptable improvement by using all components.

As mentioned above, the predictability is almost certain to be better if longer operators are used. With the above method of solution the

increase of operator length becomes impossible because the machine core is rapidly used up and significant additional time is needed for the computation. Therefore another method must be applied to obtain the longer operators or the idea of prediction must be discarded as impractical. Such a method does, however, exist and is treated in the next section, 2.2.



Figure 2.1.1 Concept Behind Least Squares Prediction Operator Experiments.











Figure 2.1.6



#### 2.2 Prediction and Spectrum Factorization

### Comparison of Prediction Techniques

We have seen in the last section that the mean squared error technique was not a practical method of prediction in the form in which it was used because of the large amount of computer space and time required. The program for prediction using the mean squared error technique was written almost entirely in FORTRAN and, due somewhat to the inefficiency of FORTRAN, the time required to obtain a 60 term self-prediction operator was about 10 minutes on the IBM 7090. The spectrum factorization method requires the spectrum as an input but the time needed to compute a 500 term wavelet is only 2 minutes on the 7090. Since the timing of both methods increases as the cube of the operator length, it is easy to see that there are tremendous advantages to the spectrum factorization method. The computation of the complete transient autocorrelation of 3000 data points and Daniell spectrum of 500 terms takes only about 2 minutes if high speed techniques are used (Simpson et al, 1961b). The Levenson (1949) technique has been programmed for the 709-7090 computers by Ralph Wiggins, but the work presented here was done before this program was available. The timing of the Levenson technique program increases as the square of the operator length but is about the same as the spectrum factorization program for a 500 term operator. The factorization method yields the minimum phase wavelet from which, as we shall see, the percent reduction can be obtained directly. The Levenson technique, on the other hand, gives the prediction operator directly, and we must compute this operator for unit prediction distance and invert it to obtain the wavelet. The choice

between the two methods might well depend on whether one wants to actually do prediction or just find the percent reduction. An iteration technique for the multi-dimensional problem has been worked out by E. A. Robinson (personal communication), and it will be quite a bit faster than the threedimensional technique described in the last section. The program for this has not been completed at the time of this publication.

#### **Decomposition**

The spectrum factorization method is much more fruitful than the mean squared error technique and the theory behind it is intimately related to the contents of section 1.4. In that section we showed that we could consider microseismic noise as a stationary ergodic time series and that, with a few additional considerations, we could assume that microseisms were generated by a white light (essentially independent) series convolved with a minimum phase wavelet. The importance of the minimum phase wavelet is that it is one sided, and therefore the expression for the present value of  $X_t$ , the microseismic noise, involves only the past values of  $\zeta_t$ the white light series. That is

$$
X_t = \sum_{i=0}^{\infty} b_i \xi_{t-i}
$$

where  $b_i$  is the minimum phase wavelet. We have seen that if  $b_i$  is known we can easily find  $\mathcal{A}_i$ , the inverse minimum phase wavelet and can therefore write

$$
\xi_{t} = \sum_{i=0}^{\infty} a_{i} \chi_{t-i}
$$
 (2.2.1)

so that all the past  $\zeta_t$  can be found from all the past  $X_t$  . We can therefore evaluate the expression for the minimum error for the mean squared error criterion (Robinson, 1954).

The minimum error is

$$
\mathbf{T}_{\mathsf{min}} = \mathbb{E} \left( \mathsf{X}_{\mathsf{t} + \mathsf{k}} - \mathsf{\hat{X}}_{\mathsf{t} + \mathsf{k}} \right)
$$

where  $X_{t+k}$  is the true value of the series at time  $t+k$ ,  $X_{t+k}$ is the predicted value, and the  $\mathbf{E}$  means expected value. The true value is, from the above considerations,

$$
X_{t+\kappa} = \sum_{i=0}^{\infty} b_i \xi_{t+\kappa-i}
$$
 (2.2.2)

But we know  $\int f(t) dt$  from equation (2.2.1), so that the error in prediction must result from our lack of knowledge of  $\xi_{t,j}$  from  $j=0$  to  $K \cdot$  Since  $\begin{cases} t \end{cases}$  are uncorrelated the best prediction we can do for them is to predict their mean, which is zero. Hence, our best prediction of  $X_{t+k}$ ,  $\hat{X}_{t+k}$ , is given by equation (2.2.2) with  $\sum_{i=0}^{k} t + k - i = 0$ for  $t+\kappa-i$ , That is

$$
\widehat{X}_{t+k} = \sum_{i=k}^{\infty} b_i \xi_{t+k-i}
$$

This has been shown to be true by Wold **(1938),** (Robinson, 1954).

# Minimum Error and Percent Reduction in Terms of the Wavelet

The minimum error is, therefore,

$$
I_{min} = E\left[\sum_{i=0}^{\infty} b_i \zeta_{t+k-i} - \sum_{i=k}^{\infty} b_i - \kappa \zeta_{t+k-i}\right]^2
$$
  
= 
$$
E\left[\sum_{i=0}^{K-1} b_i \zeta_{t+k-i}\right]^2
$$
  
= 
$$
\sum_{i=0}^{K-1} b_i^2 E\left[\zeta_t\right]^2
$$

If the expected value of  $\int_{t}^{2}$  is one

$$
T^{w_1w} = \sum_{i=0}^{k} p_i
$$

and we see that the minimum error and hence the percent reduction decreases monotonely with increasing prediction distance  $K$ . We can now easily obtain an expression for the percent reduction,  $\mathcal{R}_{\rho}$ , in terms of  $b_i$ . We recall that

$$
R_{\rm p} = 100 \left( 1 - \frac{\Gamma_{\rm min}}{\Gamma_{\rm o}} \right)
$$

where  $I_o$  is the variance of the sample,<br>  $I_o = E[X_t]^2 = E\left[\sum_{i=0}^{\infty} b_i \xi_{t-i}\right]^2$  $=\sum_{i=0}^{\infty} b_i^2$   $E[\xi_t]^2$ 

Hence

$$
\overrightarrow{R} = 100 \left( 1 - \frac{\sum_{i=0}^{K-1} b_i^2}{\sum_{i=0}^{K} b_i^2} \right)
$$

where we have made no assumptions regarding the value of  $E(\zeta_t)^2$ 

Thus we see that if  $b_i$  is known we can find the value of  $R_{\rho}$  for all  $K$  without actually computing the prediction, or even the prediction operator. We saw in section 1.4 that it is possible to find  $\mathsf{b}_i$ , and the process is called spectrum factorization. The derivation of the  $\mathbf{b}_i$ from the power spectrum is given in Appendix **E.** We see also in Appendix **E** that it is possible to find the first  $M$  terms exactly. This procedure has been programmed for the IBM **709** and **7090** computers, and the program listing, FACTOR, appears in Appendix **G.** Appendix **E** also explains most of the program logic.

We note that the expression for  $I_o$  requires all of the  $b_i$  and the program will only give us the first  $M$ . For long operators this is not troublesome since the wavelet dies off fairly rapidly. However, the estimate of  $I_0$  using just  $M$  terms will be a bit small, and therefore the value of  $R_p$  will be a bit small. We could, of course, estimate  $I_o$ from the data without using the  $\mathbf{b}_i$  since  $\mathbf{I}_o$  is just the variance,

$$
T_o = \frac{1}{N} \sum_{i=0}^{N-1} (x_i - \overline{x})^2
$$

where the mean is zero.

The computation of the minimum phase wavelet,  $\mathbf{D}_i$ , has been done for **500** terms and the corresponding percent reductions are shown in Figures 2.2.1 to **2.2.6.** Included also are some of the minimum phase wavelets and some of the inverse wavelets (Figures 1.4.1 to 1.4.5). The minimum phase wavelets for all the records are quite similar, so it is not necessary to include all of the graphs.

The percent reductions are now, of course monotonely decreasing and are forced to zero at  $t = 25$  seconds (not shown in graphs) because is computed from the first **500** terms **(25** seconds). Comparison of these figures with the self-prediction of section 2.1 (Figures 2.1.2 to 2.1.4) shows a marked increase in predictability using this technique, as much as **10** in the percent reduction, but the increase is still not large enough to improve the signal to noise ratio in the first motion interval **by** a significant amount. Comparison of the estimate of  $I_o$  from the 500 term wavelet with the sample variance estimated from **3000** data points indicates that the percent reductions obtained are off **by** less than one.



Figure 2.2.1 Percent reductions for prediction distances up to 12 seconds for records 1000, 1002, 1004.







Figure 2.2.3 Percent reductions for prediction distances up to 12 seconds for records 1006, 1008, 1010.





 $\begin{array}{c} \mathbb{Z} \\ \mathbb{Z} \end{array}$ 





 $\begin{array}{c} 1 \\ 3 \\ 6 \end{array}$ 



Figure 2.2.6 Percent reductions for prediction distances up to 12 seconds for records 1027, 1029, 1031.

### **2.3** Summary Comments on Prediction

We have seen in the last two sections that the optimum least squares prediction for short operators and for one and three dimensions are not good enough to improve the signal to noise ratio significantly. Further, we saw that the best predictions possible using the wavelet obtained **by** spectrum factorization did not yield results of any consequence. The fact that we only had **500** terms of the infinite wavelet is not important since the estimate of the standard deviation using the **500** terms was quite good (within **0.1** percent). We have alternatives of increasing the operator length of the three dimensional prediction, of going to non-linear prediction models, or, of course, of rejecting the technique of prediction of the microseisms in the first motion interval as a useful method of improving the signal to noise ratio. The first alternative, increasing the operator length for the three-dimensional case, does not seem worth trying. The improvement in predictability of the three-dimensional case, over self prediction was seen to be minescule. Further, the improvement of predictability of long operators over short was not significant. We therefore reject the first alternative.

## Independence of White Light Series

It is possible, also, to reject the second alternative, that of nonlinear prediction models. We saw, in section 1.4, in the decomposition of the microseisms to a white light series and a minimum phase wavelet, that the white light series could be considered purely random. That is, the  $\begin{cases} t \\ t \end{cases}$  were not only uncorrelated, but also statistically independent.

From elementary probability considerations we have

$$
P_{\xi_1\xi_2}(x_1,x_2) = P_{\xi_1}(x_1) P_{\xi_2|\xi_1}(x_2|x_1)
$$

The joint probability of  $\int_1^2$  and  $\int_2^2$  is equal to the marginal probability of  $\left\{\right\}$  times the conditional probability of  $\left\{\right\}$  given  $\left\{\right\}$ . If  $\begin{cases} \begin{array}{ccc} 1 & \text{if} \\ \text{if} \end{array} \end{cases}$  and  $\begin{cases} \begin{array}{ccc} \text{if} & \text{if} \\ \text{if} & \text{if} \end{array} \end{cases}$ 

$$
P_{\xi_1, \xi_2}(x_1, x_2) = P_{\xi_1}(x_1) P_{\xi_2}(x_2)
$$
   
  $\qquad \qquad ; \quad P_{\xi_2, \xi_1}(x_2 | x_1) = P_{\xi_2}(x_2)$ 

We can repeat this for many  $\zeta_i$  and obtain

$$
P_{\{n+1|\xi_1\xi_2...\xi_n}(x_{n+1}|x_{1},x_{2},...,x_{n})=P_{\{n+1}(x_{n+1})}
$$

Thus from the definition of independence we see that the knowledge of  $\left\{\bigcup_{i=1}^{n} \sum_{j=1}^{n} x_{j} \right\}$  give no information about  $\left\{\bigcap_{i=1}^{n} x_{i}\right\}$  . In a prediction problem where  $\{\zeta_{1}, \zeta_{2}, \dots \}$  are the past values and  $\zeta_{n+1}$  the future values of a time series and the  $\int_{i}^{1} \int_{i}^{1}$  to *n* are independent, we have no information about  $\int_{h+l}$  except its probability density  $P_{\int_{h+l}}(X_{n+l})$ which we kpow from the assumption of stationarity. Any prediction scheme using any of the  $\sum_{i}$   $i \neq 1$  **ro**  $\mu$  will avail us nought, but  $P_{\int \mu + i}(X_{\mu + i})$ . The best least squares prediction which one can do in the case of independence is to predict the expected value of  $\int_{n+1}^{\infty}$ , the mean, which a linear predictor can do. Therefore, if random noise can be considered as an independent white light series convolved with a minimum phase wavelet, the best prediction one can do is linear prediction, since the non-linear predictor will only bring in higher order correlations which give no new information.

Weiner (1946) states that linear prediction is optimum in the case where the noise series can be reduced to a Gaussian white light series by convolution with a operator. The reason for this can be seen from the following analysis of the joint probability density for independent and dependent variables.

## Independence and Gaussian White Light - Example

Let  $\bigcap_{i=1}^{\infty}$  and  $\bigcap_{i=1}^{\infty}$  be normally distributed independent random variables. Then the joint density of  $\int_1^{\infty}$  and  $\int_2^{\infty}$  is

$$
P_{\xi_1 \xi_2}(x_1, x_2) = P_{\xi_1}(x_1) P_{\xi_2}(x_2) = \frac{1}{2 \pi \pi \sigma_2} P_{\xi_2}(x_1) = \frac{1}{2 \pi \sigma_1^2} \frac{1}{2 \sigma_1^2} \frac{x_1^2}{2 \sigma_2^2}
$$

where  $\sigma_i$  is the standard deviation of  $\gamma_i$ . Now we define  $y_i$  and  $yz$ as a linear combination of  $X_i$  and  $X_2$ 

$$
y_1 = \alpha x_1 + b x_2
$$
  
 
$$
y_2 = c x_1 + d x_2
$$
 (2.3.1)

and therefore

or

$$
P_{\eta_1, \eta_2}(y_1, y_2) dy_1 dy_2 = P_{\xi_1, \xi_2}(x_1, x_2) dx_1 dx_2
$$
  

$$
P_{\eta_1, \eta_2}(y_1, y_2) = |J| P_{\xi_1, \xi_2}(x_1, x_2)
$$

where  $|\mathbf{J}|$ , the magnitude of the Jacobian for this transformation, is  $J = ad-bc$ 

Solving (2.3.1) for  $X_i$  and  $X_2$ :

$$
x_1 = \frac{d}{J} y_1 - \frac{b}{J} y_2
$$
  
 $x_2 = \frac{a}{J} y_2 - \frac{c}{J} y_1$ 

Hence joint density for the dependent variables  $\eta$ , and  $\gamma$ <sub>2</sub> is  $P_{\gamma_1, \gamma_2}(y_1, y_2) = \frac{1 J l}{2 \pi \sigma_1 \sigma_2} exp \left[ - \left( \frac{\sigma_1^2 d^2 + \sigma_2^2 c^2}{2 \sigma_1^2 \sigma_2^2 J^2} \right) y_1^2 \right]$  $-\left(\frac{\sigma_1^2 \alpha^2 + \sigma_2^2 b^2}{2 \sigma_1^2 \sigma_2^2 + 7^2}\right) y_2^2 + \left(\frac{\sigma_2^2 bd + \sigma_1^2 \alpha c}{\sigma_1^2 \sigma_2^2 + 7^2}\right) y_1 y_2$ 

We note the expected values of the following quantities.

$$
\mathcal{U}_{1} = E(y_{1}^{2}) = \alpha^{2} \sigma_{1}^{2} + b^{2} \sigma_{2}^{2}
$$
  

$$
\mathcal{U}_{2} = E(y_{2}^{2}) = c^{2} \sigma_{1}^{2} + \alpha^{2} \sigma_{2}^{2}
$$
  

$$
\mathcal{U}_{12} = E(y_{1}y_{2}) = \alpha c \sigma_{1}^{2} + bd \sigma_{2}^{2}
$$

Thus

$$
P_{\eta_1, \eta_2}(y_1, y_1) = \frac{|\mathcal{T}|}{2 \pi \pi \sigma_2} \exp \left[ \frac{-\mu_1 y^2 - \mu_2 y^2 + 2\mu_{12} y_1 y_2}{2 \pi^2 \sigma_2^2 \mathcal{T}^2} \right]
$$

If  $\mu_{12}$ , the correlation of  $y_1$ , and  $y_2$ , is zero, the cross term in the exponential is zero and  $P_{\eta,\eta_2}(y_1,y_2)$  factors. This can be extended for  $P_{\eta_1,\eta_2,\dots,\eta_n}(y_1,y_2,\dots,y_n)$  and we see that in general if the correlation coefficients are zero the joint density of h variables factors. Hence for the Gaussian, linear independence implies statistical independence. (Davenport and Root, 1950).

# Non-Linear Operators

We thus see the reason behind Wiener's statement that linear prediction is optimum if it reduces the series to Gaussian White light. We need actually only show, therefore, that the white light series,  $\xi_t$  is Gaussian in order to reject the adoption of a non-linear predictor. We saw in section 1.4 that, for microseisms,  $\zeta_t$  was Gaussian in many cases, and was in general nearly Gaussian. We can fall back on the independence tests for these non-Gaussian cases which showed that we could consider  $\zeta_t$ independent. The independence of  $\left\{ \right. \right.$  forces us to drop the notion of non-linear prediction and hence forces us to reject the technique of prediction for signal to noise ratio improvement in the first motion interval.
# **3. AUTOMATIC** DETECTION OF SIGNALS IN MICROSEISMIC NOISE

### 3.1 Detection System

## Description - Inputs and Outputs

A detection system to automatically detect signals in microseismic noise has been designed and a computer program has been written to simulate the system. The system and programs have been developed by **S.** M. Simpson, Jr., for Geoscience, Inc. A flow chart of the computer simulation of the system appears in Figure 3.1.1. The signal plus noise input is rectified by squaring or by taking the absolute value and this rectified waveform is averaged. The averaged rectified wave form then enters a network which decides if there is a signal present or not, and sets an alarm if there is a signal. The system variables are the type of rectification, the averaging time, the hesitation time and the alarm level. The averaging time is the length of time over which the rectified waveform is averaged before going to the decision network. Averaging over some length of time is necessary to reduce false alarms due to an occasional high noise amplitude, but the length must not be much greater than the expected length of the signal, since the average would be too small to trigger the alarm. The hesitation time is the length of time that the rectified averaged input must remain above the alarm level before an alarm is sounded. This also tends to cut down alarms which might be caused by noise spikes. The alarm level is the ratio of the value which averaged rectified wave must reach for an alarm to the r.m.s. amplutide of the noise.

It is, therefore, the signal to noise ratio at which the system can operate. For example, if the alarm level is 1.75, an alarm will not be sounded until the average rectified waveform reaches 1.75 times the r.m.s. noise amplitude.

The system as it stands is an event detector. It tells whether or not an event has occurred, but makes no statement as to the nature of the signal which triggered the alarm. Such a system could be used in an automatic nuclear surveilance network to control the collection of data. Only data near the time of an alarm would be recorded, and these alarms could be studied for source type. An alternate procedure would be to collect all data and just study the portions corresponding to alarms.

In order to rate the effectiveness of this system, it is necessary to study the false alarm rate and failure to detect rate as a function of the system parameters. The next few sections give the results of false alarm and failure rate studies on the computer simulated system for raw and filtered signals and noise.



Figure **3.1.1** Computer Simulation Flow Chart

### **3.2** False Alarm Rate - FALARA

#### Generation of Input Noise

The false alarm rate of the detection system can be obtained by using a pure noise input rather than a signal plus noise input and counting the number of times an alarm is sounded as a function of the system parameters. A large amount of noise representing many hours of sequencial microseisms is necessary to carry out the study. Since only a few minutes of consecutive microseismienoise is available from our digitized noise library, the microseisms must be generated artificially. We have seen in section 1.4 that this could be done to a good approximation using a minimum phase wavelet from real data and Gaussian white noise. Thus, the artificial microseisms,  $\lambda_t$ , shown in the upper trace of Figure 1.4.16, are generated by the convolution

$$
X_t = \sum w_i \, \zeta_{t-i}
$$

where  $w_i$  is the wavelet and  $\{t\}$  is the Gaussian white noise. The wavelet used in these studies was computed from record 1002, the vertical component of the noise before the Logan shot 1902 km from the shot point. The Gaussian white noise is generated from the Rand random digits by summing non-overlapping groups of ten digits. The central limit theorem tells us that the resulting sequence will have an approximately normal distribution.

A 500 term minimum phase wavelet was computed and every other point was then deleted. This left a 250 point wavelet with an equivalent

digitization rate of 10 points per second. The deletion is not unreasonable since there is almost no power above **5** cps. This wavelet was then convolved with 85,249 points of Gaussian white noise to yield 85,000 points of artificial microseisms which correspond to 2.22 hours of noise.

# False Alarm Rate Studies

The computer program FALARA (FAlse Alaram RAte) has been written by S. M. Simpson to simulate the detection system with pure microseismic noise input. For each set of system parameters the simulation was continued until either 100 alarms were sounded or all 85,000 points of noise were used. A flow chart of the simulation for the false alarm rate is shown in Figure 3.2.1 along with the system parameters used. As can be seen from this figure, two different types of rectification were used with five averaging times, ten alarm levels and five hesitation times. The false alarm rate is computed in units of alarms per hour. The results are shown in Figures **3.2.2** and 3.2.3 where the false alarm rate is plotted against the alarm level for several averaging times and for both types of rectification. Each figure is for a different hesitation time. Curves are included for only part of the results, but these are sufficient to indicate over-all trends in the system.

It is obvious that a desirable system should have very few false alarms for a low alarm level. We see from the figures that the curves with both low false alarm rate and low alarm level are relatively insensitive to hesitation time. For a given hesitation time the curves show that a long averaging time is desirable. These qualitative results are just as expected. The noise amplitudes change fairly rapidly and the

high noise values, which are of short duration, are what trigger the alarm. Consequently the curves for short averaging time are affected by the hesitation time whereas the curves for long averaging time are only slightly changed. We note that for given averaging and hesitation times the curves for rectification by squaring are always better. We also see that the curves for high averaging times are fairly close together, which indicates that very little improvement will be obtained with averaging times greater than 10 seconds.



Figure 3.2.1 False Alarm Rate Flow Chart





#### **3.3** Failure Rate - FAILRA

#### Description of System

The failure rate of the detection system is somewhat more difficult to obtain than the false alarm rate. Both signal and noise are required along with several signal to noise ratios. In the simulation of the system, the signal, scaled to give the required r.m.s. signal to noise ratio, and a block of noise are added together to give the input waveform. This is rectified and averaged and sent to the decision network where the alarm is announced if triggered. Figure **3.3.1** shows a flow chart of the computer program FAILRA (FAILure RAte), written **by S.** M. Simpson, with the system parameters used to obtain the failure rate.

The artificial microseismic noise used for the false alarm rate determination was used for the failure rate studies. For the signal it was necessary to pick out a representative bomb record with a fairly high signal to noise ratio so that the noise occurring with the signal was negligible compared to the microseismic noise added later. The record chosen was the vertical component of the signal from the Blanca shot recorded at **1398** km from the shot point (record **58,** see Figure **3.3.2).** Every other point of the first **600** points of this record were used thus giving **30** seconds of signal. The signal to noise ratios used were **1.78, 2.07, 2.37, 2.67, 2.97, 3.26, 3.56,** 4.0, 4.45 and 5.34.

# Failure Rate Studies

The system simulation was carried out for a hesitation time **1.5** seconds, both types of rectification, five averaging times, ten alarm

levels and all above signal to noise ratios. For each set of system parameters the detection was tried 101 times and the number of successes and failures noted. In graphs showing the results, Figures 3.3.2 and 3.3.3, the success probability is plotted against alarm level for different averaging times. Each figure gives the curves for a different signal to noise ratio. The complete set of results is not given since the success probabilities for signal to noise ratios greater than 3.26 are nearly all equal to one.

The curves show that the long averaging times are successful over a smaller range of alarm levels than the short averaging times for a given signal to noise ratio, and they stop being successful at an alarm level approximately equal to the signal to noise ratio. This is not surprising since the long averaging time will average the signal alarm but the short averaging time will permit high amplitude pulses to trigger an alarm.

The wider range of success for short averaging times is offset by the unavoidably large false alarm rate which was noted in the last section. The most generally effective system parameters must balance the false alarm rate and the failure rate. In Figure 3.3.4 the overall system effectiveness, taking into account both false alarms and failures, is shown as a graph of signal to noise ratio versus false alarm rate for .95 success probability. The curves were obtained, for a given averaging time, by picking off the alarm levels for .95 probability of success for all signal to noise ratiosand then turning to the false alarm rate curves and picking the false alarm rates for the previously obtained alarm levels. The

hesitation time was kept at 1.5 for these curves. We see that, for smaller signal to noise ratios, rectification by squaring and use of long averaging times are best. For a signal to noise ratio of 1.78 and 10 second averaging time gives about 10 false alarms per hour, and as the signal to noise ratio increases the false alarm rate drops sharply so that the system is quite good at high signal to noise ratios. The large number of false alarms make the system relatively ineffective for signal to noise ratios less than 1.78.



Figure 3.3.1 Failure to Detect Flow Chart







# 3.4 Automatic Detectior with Filtering

# Band Pass Filters and the Signal to roise Ratio

The last section showed the overall effect of the detection system and indicated that it was not particularly good for signal to noise ratios less than **1.78.** If, however, the signal to noise ratio of the raw data can be improved by filtering, the usefulness of the detection system may be increased enormously. Examination of the spectra of the noise records (Figures **1.3.6** to **1.3.9)** show that most of the power is between **0** and about **.7** cps with a few spikes around 1.4 and 2.0 cps. The vertical records have less energy at the higher frequencies than do the horizontals. If we look at the noise spectra through a window from **.7** to **1.8** cps we see only a very small percentage of the total power. The signal, on the other hand, has energy all through this band. If a reasonable percentage of the total signal power appears in this range of frequencies, a simple band pass filter will improve the signal to noise ratio quite a bit.

The programs FAILRA and FALARA can be used again to study the failure and false alarm rates **by** pre-filtering the signal and noise and the proceeding as in the last two sections. The flow charts in Figures **3.2.1** and **3.3.1** are applicable if "Noise Tape" is changed to "Filtered Noise Tape", and "Signal Tape" changed to "Filtered Signal Tape."

The signal to noise ratio improvement obtained by band pass filtering can be estimated from the spectra of the signal and the noise which are shown in Figure 3.4.1. If the signal and noise were initially scaled to have a one-to-one ratio, and were then band pass filtered to pass **.8** to **1.7** cps

we see that nearly all the signal would remain and nearly all the noise would be removed. The signal to noise ratio improvement for this case would be a factor of about 5.

#### Effect of Filter on System Characteristics

It is important to see if the detection system characteristics change significantly when the filtered signal and noise both have band widths which are narrow compared to the band widths of the raw signal and noise. If the characteristics are relatively invariant with band width, the system can be said to be an energy detector and its effectiveness can be measured in terms of the signal to noise ratio improvement brought about by the filtering, and the system response to unfiltered signals.

The constancy of the system to change in band width was studied by band pass filtering the signal and noise separately and using the programs FAILRA and FALARA to obtain the false alarm rates and failure rates. The signal to noise ratios and alarm levels were computed from the amplitudes of the filtered noise and signal. The results of the study are shown in Figures 3.4.2 to 3.4.6. As in the last two sections, the false alarm rate is shown as a graph of the number of false alarms per hour against alarm level, the failure rate is given by the success probability as a function of alarm level, and the system's effectiveness is shown in a graph of the false alarm rate versus signal to noise ratio. In comparing these graphs to the ones for unfiltered data we see only slight differences. The trends are all the same and the actual curves, particularly those for longer averaging time, are approximately the same. The overall system effectiveness is also about the same for the filtered and unfiltered cases.

In view of the findings from the filtered and unfiltered cases we can say that the system is essentially an energy detector and that the curves obtained for the unfiltered case can be used for the filtered case if we can compute the signal to noise ratio improvement due to filtering. We have seen that for the particular signal and noise used this improve- .ment was enormous and results in an extremely low false alarm rate. With the use of the curves which have been presented one can easily compute the range of signal amplitudes which can be detected reliably if the level of the background noise is known.



Figure 3.4.1 Signal and Noise Auto Spectra











## 4. SUMMARY

The seismic data from the Logan and Blanca underground nuclear shots, which was provided by the Air Force, has been digitized and, along with other data contributed **by** Dr. Bruce Bogert and by United Electro Dynamics, Inc., has been subjected to many computational experiments. In the first of these the microseism data was considered as a signal and the object was to infer the nature of the sources and the wave types involved. We saw that the amplitude of the microseisms at about .3 cps decreased with increasing distance from the coast, but the higher frequency did not display any regular trend. The suggestion is that the low frequency noise is of oceanic origin whereas the higher frequencies are more likely of local origin. It was not possible to pin down Rayleigh and Love waves with any degree of certainty, but their presence was not disproved. The failure of the wave type experiments is attributed to the complex nature of the microseisms. The model used cannot deal with many waves of the same frequency but different directions of travel.

The inadequacy of a simple deterministic model motivated a statistical treatment of microseismic noise. The microseisms are considered as a time series and, under the ergodic hypothesis, the relative constancy of the power density spectrum suggests that the time series is at least wide sense stationary. Studies on the microseism amplitudes show that their probability distribution is Gaussian and that they are dependent.

The power density spectra have been computed using the Daniell technique. The spectra are quite similar in structure over distances of

several hundred kilometers. There is a prominent peak at about **.3** cps and in some cases there are peaks at 1.4 and 2 cps. The low frequency peak is interpreted as the high end of the oceanic microseism band which is cut off on the low end **by** the seismometer response. The higher frequencies are attributed to local causes.

Cross spectra of different components at the same station, like components from different stations, and array data have been computed. Again it is difficult to pick out individual wave types and it is not possible to follow waves from one station to another. This is again attributed to the complex structure of the noise.

Since the microseisms can be considered as a wide sense stationary time series, a mathematical description is possible. The moving summation and autoregressive representations are valid. With the assumption of an absolutely continuous spectral density the spectra can be factored and a minimum phase wavelet found for the moving average representation. The generating model for microseisms is then a white light series into a minimum phase system. Probability studies on the white light series obtained **by** convolving the inverse minimum phase wavelet with the original data show that the white light is essentially Gaussian and independent.

The minimum phase wavelet is also the predictive decomposition and can be used to compute the predictability of the microseisms. This technique of prediction is found to be faster and easier to handle than the mean aquare error method, although the Levinson technique is quite good. The predictability of the microseisms is not very great. About half the energy **(50** percent reduction) can be predicted for one or two seconds and then the

decrease is fairly rapid. Multidimensional prediction does not give appreciably better results than the one dimensional or self prediction. Thus prediction as a method of noise reduction in the first motion interval is not particularly promising. We can say, however, that our linear prediction is the best we can do, and that non-linear operators will not help. This is because the microseisms can be considered to be generated by Gaussian white noise into a minimum phase system. In this case the white noise is independent and higher correlations give no information about the noise.

The mathematical model enables us to generate artificial microseisms so that long periods of continuous noise are- available. These long noise series are required by the computer program which simulates a detection system. Noise above is needed to compute the false alarm rate and signal plus noise is needed for the failure rate. The system effectiveness is plotted on a graph of false alarms per hour as a function of signal to noise ratio for **95%** detection probability **(5%** failure rate). The system characteristics are found to remain approximately constant when a band pass filter is introduced at the input. Thus the system will function as an energy detector and band pass filters can be used to improve the signal to noise ratio. Improvement of a factor of five was found for the particular signal, noise, and filter used.

The emphasis has been on the statistical approach throughout this thesis. There is, of course, plenty of room for additional work of both statistical and deterministic nature on the available data in the same general area as the present work. More complicated models which take into account several wave types and many directions of travel may be

introduced and fitted to the data. New techniques will enable multidimensional prediction studies with long operator lengths, and it would be interesting to compare results of this sort of study with the long operator studies of section 2.2.

The cross correlation results on the array data certainly do not represent exhaustive study. Multi-dimensional prediction experiments as well as summation of records with variable time lags would be quite interesting. Three component and array detection system studies by computer simulation would also prove useful.

## APPENDIX A

#### WATER WAVE PROBLEM

Longuet-Higgins (1950) has shown that a standing wave can produce a second order pressure fluctuation which is unattenuated with depth and which has twice the time frequency of the standing wave.. Hence it is possible to show that microseisms could be produced in deep water even though the linear theory tells us that the pressure fluctuations die off exponentially with depth. In order that there be enough energy transmitted to the bottom, there must be a "patch" of standing waves which is coherent over a fairly large area and the patch must not move because the motion will cause the pressure oscillations to average out to zero. Therefore the standing waves must meet nearly head on. In fact, it has been shown (Kenyon, 1961) that if the travelling waves meet at an angle  $\theta$  ( $\theta = 0$ , head on), the average pressure on the bottom must be multiplied by  $exp(-2h \times sin\theta)$ where  $h$  is the depth of the water,  $\alpha$  the wave number and  $\Theta$  the angle between the travelling wave fronts.

There is a special case of interest when the waves meet at such an angle that the "patch" of standing waves moves with a velocity,  $\bigvee_{S}$ , equal to the velocity of propagation of Rayleigh waves,  $V_{r}$ , in the medium. The travelling waves, with velocity  $V_t$ , must meet at an angle  $\Theta$ such that

$$
V_t = V_r \big/ \sin(\theta / a)
$$

In this case there is essentially a resonance and strong microseisms

could build up if the "patch" of water waves remains coherent for a long enough time.

One of the problems considered by Longuet-Higgins was the two dimensional compressible case of a layer of water with a rigid lower boundary and a standing wave at the surface. His solution requires the small parameter expansion technique of handling non-linear problems and illustrates the frequency doubling effect as well as organ pipe resonance. The problem which will be treated here is a good deal simpler in that it considers the incompressible transient problem. This is done to illustrate the energy swapping to the sum and difference frequencies of all frequencies present and uses a representation for non-linear problems devised by DeVorkin (1963). DeVorkin's scheme is particularly useful in that the solution is in terms of kernels which do not depend on the initial conditions. Therefore once the kernels have been found for a given geometry the solution of many problems with different initial conditions can readily be found. The method is also useful for statistical initial conditions.

We consider the two dimensional transient problem of an incompressible irrotational fluid layer of constant thickness,  $\boldsymbol{h}$  , over a rigid half space with arbitrary initial conditions on the velocity and surface shape. We assume a velocity potential  $\varphi$ . The velocity is therefore  $\vec{v} = -\vec{\nabla}\phi$ . The continuity equation is then  $\nabla^2 \psi = 0$  and the equation of motion is

$$
\frac{\partial \vec{v}}{\partial t} + (\vec{v} \cdot \vec{\nabla})\vec{v} + g \vec{\hat{\Lambda}} + \frac{1}{\rho} \vec{\nabla}P = 0
$$

where  $\int$  is the gravitational potential,  $\int$  is the density (assumed constant) and  $\beta$  the pressure. We factor out a  $\nabla$  and obtain Bernoulli's equation

$$
-\frac{\partial f}{\partial \phi} + \Delta \phi \cdot \Delta \phi + \frac{\partial f}{\partial \phi} + \frac{\partial f}{\partial \phi} = 0
$$

where  $\chi$  is negative downward and  $p = o$  at the surface  $\chi \cdot \eta$ . The free surface condition is

$$
\frac{\partial \varphi}{\partial x} = \frac{\partial \eta}{\partial x} - \frac{\partial \varphi}{\partial y} + \frac{\partial \eta}{\partial t} = 0 \quad \text{at} \quad y = \eta(x, t) \quad (A-1)
$$

Bernoulli's equation becomes at  $\chi$ =*C* 

$$
-\frac{\partial \varphi}{\partial t}\Big|_{\delta=\gamma}+\left(\frac{\partial \varphi}{\partial x_{i}}\right)^{2}\Big|_{\delta=\gamma}+\left(\frac{\partial \varphi}{\partial \delta}\right)^{2}\Big|_{\delta=\gamma}+\varphi\gamma=0
$$
 (A-2)

The solution to the continuity equation which satisfies the condition  $\rightarrow$  = 0 at  $\lambda$  = -h is (A-3)  $\psi(x, y, t) = \sum_{m=1}^{M} \Phi_{m}(t) \left[ e^{-m}t + e^{amh}e^{m}t \right] e^{-imx}$ 

where we have assumed a discrete set of frequencies. DeVorkin's representation scheme applies to total differential equations and hence to the Fourier transform over the spacial frequencies of the boundary equations.

The initial conditions are

for 
$$
\varphi
$$
:  $F(m, 0)$ ,  $m = -M$  to  $M$   
for  $\eta$ :  $N(m, 0)$ ,  $m = -M$  to  $M$ 

where  $F(m,t)$  and  $N(m,t)$  are the Fourier transforms of  $\ell(n)$  and  $\gamma(1)$ . We combine these into a single variable

$$
\mathcal{V}_{\alpha} , \alpha = 1, ..., 4m+2 \qquad where
$$

$$
\psi_{1} = F(-m, o), \psi_{2} = N(-m, o), \psi_{3} = F(-m+v, o), \psi_{4} = N(-m+v, o), etc.
$$

The representation scheme is then:

 $\ddot{\phantom{a}}$ 

$$
F(m,t)=\sum_{\alpha}K_{\alpha}^{m}H_{\alpha}^{l}+\sum_{\alpha\beta}K_{\alpha\beta}^{m}H_{\alpha}^{l}H_{\beta}+\sum_{\alpha\beta\gamma}K_{\alpha\beta\gamma}^{m}H_{\alpha}^{l}H_{\beta}H_{\gamma}+...
$$
  

$$
N(m,t)=\sum_{\alpha}L_{\alpha}^{m}H_{\alpha}+\sum_{\alpha\beta}L_{\alpha\beta}^{m}H_{\alpha}^{l}H_{\beta}+\sum_{\alpha\beta\gamma}L_{\alpha\beta\gamma}^{m}H_{\alpha}^{l}H_{\beta}H_{\gamma}+...
$$

which can be combined to

$$
\psi_n(t) = \sum_{\alpha} R_{\alpha}^{n} \psi_{\alpha} + \sum_{\alpha \beta} R_{\alpha \beta}^{n} \psi_{\alpha} \psi_{\beta} + \sum_{\alpha \beta \gamma} R_{\alpha \beta \gamma}^{n} \psi_{\alpha} \psi_{\beta} \psi_{\gamma} + \cdots \quad (A-4)
$$

where

$$
\Psi_{n}(t) = F(\frac{n-2m-1}{2},t)
$$
 for nodd, 21  
 $\Psi_{n}(t) = N(\frac{n-2m-2}{2},t)$  for never, 22 (A-5)

The R's are thus system functions independent of initial conditions.

The boundary equations  $(A-1)$  and  $(A-2)$  apply at  $\gamma : \gamma$  but since

is unknown the equations must be expanded in a Taylor series about **-0** in powers of **.** Expanding to second order only

$$
-\oint_{t} -\frac{\partial \oint_{t}}{\partial \gamma} \gamma + \oint_{x}^{2} + \oint_{\delta}^{2} + \gamma \gamma = 0
$$
 (A-6)

$$
\psi_{x} \gamma_{x} - \psi_{\overline{\delta}} - \frac{\partial \varphi_{\overline{\delta}}}{\partial \overline{\delta}} \eta - \eta_{\overline{\epsilon}} = 0
$$
 (A-7)

where the subscripts denote differentiation.

We take the Fourier transform of these equations to obtain  
\n
$$
\vec{F}(m) = \hat{i} \sum_{p} p C(p) \vec{F}(p) N(m-p) - \sum_{p} p F(p) (m-p) F(m-p) +
$$
\n
$$
- \sum_{p} p C(p) F(p) (m-p) C(m-p) F(m-p) + q N(m) = 0
$$
\n(A-8)

for equation (A-6) and a similar expression for equation **(A-7).** In this transformation we have used the fact that multiplication in one domain is convolution in the other, and have set the transform of  $\frac{\partial \varphi}{\partial \chi}$ equal to  $C(m)$   $\mathsf{F}(m)$  . The dots represent time differentiation. We note that equation **(A-8)** contains more than one term with a time derivative. Poincare's theorem on small parameter expansions does not guarantee a solution unless the right-hand side contains not time derivatives. We can, however, consider all the time derivative terms as an operator,  $H$ , operating on  $F(m)$  and then show that the operator  $H = \int f \cdot \alpha$  can be inverted if  $\alpha$  is small. That is, if the operator H cannot in general be inverted, we must demand that it can be expressed

as  $I - Q$  where  $Q$  is small enough that the Neumann series resulting from the inversion converges. Hence, for many cases we must impose the restriction that the non-linear terms be small compared to the linear ones.

Since  $H$  can be inverted we go ahead and use the representation scheme equating terms of like order and remembering that the equations must hold for arbitrary initial conditions,  $\psi_{\alpha}$ 

The first order equations are from equations  $(A-6)$ ,  $(A-7)$  and  $(A-8)$ , using the notation introduced in equations  $(A-4)$  and  $(A-5)$ ,

$$
\dot{R}_{\alpha}^{n+1} + \left(\frac{n-1}{2}\right) C\left(\frac{n-7}{2}\right) R_{\alpha}^{n} = 0 \qquad j \qquad J=2M+1
$$

These can be solved to give

$$
R_{\infty}^{h} = Q_{+} \exp[i \gamma(n, \texttt{J}) t] + Q_{-} \exp[-i \gamma(n, \texttt{J}) t]
$$

for *n* odd, where

$$
\gamma(n, J) = \sqrt{2\left(\frac{n-1}{2}\right)} C\left(\frac{n-1}{2}\right)
$$
  
Q<sub>+</sub> = 
$$
\frac{2\gamma(n, J)}{2\gamma(n, J)} \int_{n\alpha}
$$
  
Q<sub>-</sub> = 
$$
\frac{-2\gamma(n, J)}{2\gamma(n, J)} \int_{n\alpha}
$$

where  $\int_{N_N}$  is the Kronecker delta, and

$$
R_{\alpha}^{n+1} = b_+ \exp[i \delta(n,\tau) t] + b_-\exp[-i \delta(n\tau) t]
$$

for h odd, where

$$
b_{+} = \frac{-\frac{n-1}{2}C(\frac{n-3}{2}) - \gamma(n, T)}{2 \gamma(n, T)} \delta_{n+1, \alpha}
$$

b<sub>-</sub> = 
$$
\frac{h-T}{2} \frac{\Gamma(\frac{n-1}{2}) + \gamma(n, \tau)}{2 \gamma(n, \tau)}
$$

The above equations for  $R_{\alpha}^{n}$  and  $R_{\alpha}^{n+1}$  are correct for  $n \neq \mathcal{T}$  . For  $n = J$ ,  $R_N^n$  and  $R_N^{n+1}$  are zero for all  $t$ .

The second order equations, obtained **by** equating the second order terms in equations **(A-6), (A-7)** and **(A-8)** containing the second order kernels and convolutions of the first order terms. The convolutions may easily be performed and the  $R_{\alpha\beta}^{n}$ ,  $R_{\alpha\beta}^{n+1}$  equations can be considered as a matrix equation. However, due to the simple coupling of the equations only a  $2 \times 2$  matrix need be considered. The zero spacial frequency, **1=** J , must again be considered as a special case.

The second order equations are

$$
\sum_{k\in\mathbb{N}} \dot{R}_{k}^{n} e^{i\theta_{k}} \psi_{k} \psi_{l} - \frac{1}{2} \sum_{k\in\mathbb{N}} R_{k}^{n+1} \psi_{k} \psi_{l} = \sum_{p=1}^{N-1} \frac{p-1}{2} \alpha \left( \frac{p-1}{2} \right) \sum_{k\in\mathbb{N}} \dot{R}_{k}^{p} \psi_{k} \sum_{k\in\mathbb{N}} R_{k}^{n-p} \psi_{k}
$$
\n
$$
+ \sum_{p=1}^{N-1} \left( \frac{n-1}{2} \right) \left( \frac{n-p}{2} \right) \left( 1 - C \left( \frac{n-p}{2} \right) \right) C \left( \frac{n-1}{2} \right) \sum_{k\in\mathbb{N}} \beta_{k}^{p} \psi_{k} \sum_{k\in\mathbb{N}} R_{k}^{n-p+1} \psi_{k} \tag{A-9}
$$
where  $N: 4M + 2$  and  $N$  and  $P$  are odd,

$$
\sum_{\kappa} \hat{R}_{\kappa}^{n+1} \psi_{\kappa} \psi_{\epsilon} + \frac{n-1}{2} C\left(\frac{n-1}{2}\right) \sum_{\kappa} R_{\kappa}^{n} \psi_{\kappa} \psi_{\epsilon} =
$$

The equations must hold for arbitrary  $\frac{1}{N}$  so that

$$
\hat{R}_{\kappa}^{n} - q \cdot R_{\kappa}^{n+1} = \sum_{p=1}^{N-1} \left( \frac{p-1}{2} \right) C \left( \frac{p-1}{2} \right) \left[ \dot{R}_{\kappa}^{p} R_{\ell}^{n-p} + \dot{R}_{\ell}^{p} R_{\kappa}^{n-p} \right] +
$$
\n
$$
+ \sum_{p=1}^{N-1} \left( \frac{n-1}{2} \right) \left( \frac{n-p}{2} \right) \left[ -C \left( \frac{n-p}{2} \right) \right] C \left( \frac{p-1}{2} \right) \left[ R_{\kappa}^{p} R_{\ell}^{n-p+1} + R_{\ell}^{p} R_{\kappa}^{n-p+1} \right]
$$
\n(A-10)

and

and  
\n
$$
\hat{R}_{\kappa q}^{n+1} + (\frac{n-1}{2}) C(\frac{n-1}{2}) \hat{R}_{\kappa q}^{n} = - \sum_{p=1}^{N-1} (\frac{p-1}{2}) (\frac{n-1}{2}) [\hat{R}_{\kappa}^{p} \hat{R}_{p}^{n-p} + \hat{R}_{q}^{p} \hat{R}_{\kappa}^{n-p}]
$$

The convolutions are not hard since  $R_{\alpha}^{h}$  is diagonal. The last two equations may be written

$$
\dot{R}_{\kappa\ell}^{n} - q R_{\kappa\ell}^{n+1} = T_{\kappa\ell}^{n}
$$
\n
$$
\dot{R}_{\kappa\ell}^{n+1} + \left(\frac{n-1}{2}\right) C\left(\frac{n-1}{2}\right) R_{\kappa\ell}^{n} = T_{\kappa\ell}^{n+1} \qquad n \text{ odd}
$$

We write this as a matrix equation

$$
\begin{bmatrix}\n\hat{R}_{\kappa\ell}^{n} \\
\hat{R}_{\kappa\ell}^{n+1}\n\end{bmatrix} + A \begin{bmatrix}\nR_{\kappa\ell}^{n} \\
R_{\kappa\ell}^{n+1}\n\end{bmatrix} = \begin{bmatrix}\nT_{\kappa\ell}^{n} \\
T_{\kappa\ell}^{n+1}\n\end{bmatrix}
$$

where  $A$  is the matrix

$$
A = \begin{bmatrix} 0 & -\frac{3}{4} \\ (\frac{n-1}{2})C(\frac{n-1}{2}) & 0 \end{bmatrix}
$$

The solution to the equation is, then,

$$
\begin{bmatrix} R_{\kappa\ell}^{\eta} \\ R_{\kappa\ell}^{\eta+1} \end{bmatrix} = \int_{0}^{1} e^{-A(t-\tau)} \begin{bmatrix} T_{\kappa\ell}^{\eta} \\ T_{\kappa\ell}^{\eta+1} \end{bmatrix} d\tau
$$

Since  $R_{K\ell}^{n}$ ,  $R_{K\ell}^{n+1}$  = 0 at  $\tau$  = 0. This is simplified considerably if  $A$  can be diagonalized. If  $\bigcup$  is the transformation matrix for this diagonalization then  $R_{\kappa\downarrow}^n = \bigcup \mathcal{S}_{\kappa\downarrow}^n$ and

$$
U\begin{bmatrix} \dot{S}_{\kappa\ell}^n \\ \dot{S}_{\kappa\ell}^n \end{bmatrix} + AU\begin{bmatrix} S_{\kappa\ell}^n \\ S_{\kappa\ell}^n \end{bmatrix} = \begin{bmatrix} T_{\kappa\ell}^n \\ T_{\kappa\ell}^n \end{bmatrix}
$$

multiplying by U<sup>-1</sup>  

$$
\begin{bmatrix} \dot{S}_{r,l}^n \\ \dot{S}_{r,q}^{n+1} \\ \dot{S}_{\kappa\ell}^{n+1} \end{bmatrix} + U^{-1}A U \begin{bmatrix} S_{\kappa\ell}^n \\ S_{\kappa\ell}^{n+1} \\ \dot{S}_{\kappa\ell}^{n+1} \end{bmatrix} = U^{-1} \begin{bmatrix} T_{\kappa\ell}^n \\ T_{\kappa\ell}^{n+1} \\ T_{\kappa\ell}^{n+1} \end{bmatrix}
$$

where  $U^{-1}AU = D$ is diagonal.

Then

$$
\begin{bmatrix} S_{\kappa 1}^n \\ S_{\kappa \ell}^{n+1} \end{bmatrix} = \int_{0}^{\mathbf{t}} e^{-D(t-\tau)} U^{-1} \begin{bmatrix} T_{\kappa \ell}^n \\ T_{\kappa \ell}^{n+1} \end{bmatrix} d\tau
$$

and

$$
\begin{bmatrix} R_{\kappa\ell}^{n} \\ R_{\kappa\ell}^{n+1} \end{bmatrix} = \int_{0}^{\tau} \bigcup e^{-D(t-\tau)} U^{-1} \begin{bmatrix} T_{\kappa\ell}^{n} \\ T_{\kappa\ell}^{n+1} \end{bmatrix} d\tau
$$

For the matrix AU and U<sup>-1</sup> are  
\n
$$
U = \begin{bmatrix}\n1 & 1 \\
-\frac{1}{2}\sqrt{\frac{(n-3)(-\frac{n-3}{2})}{3}} & \frac{1}{2}\sqrt{\frac{(n-1)(-\frac{n-3}{2})}{3}} \\
1 & -\frac{1}{2}\sqrt{\frac{(n-1)(-\frac{n-3}{2})}{3}} \\
1 & \frac{1}{2}\sqrt{\frac{(n-3)(-\frac{n-3}{2})}{3}} \\
1 & \frac{
$$

where

$$
X = exp(i \gamma(n)(t \cdot \tau))
$$
  

$$
Y = exp(-i \gamma(n)(t \cdot \tau))
$$

For the zero spacial frequency, which is the frequency of interest for deep water microseism generation,  $n: J, J \leftarrow \mathbb{I}$ , we have from equation **(A-11)**

$$
\dot{R}_{\kappa\ell}^{\text{JH}} = 0 \quad , \qquad R_{\kappa\ell}^{\text{JH}} = 0
$$

In equation  $(A-10)$  we note a symmetry in  $k$  and  $\ell$  so that we need only consider half of the right-hand side from which we determine half the solution for  $R^T \kappa$ **!** . We call this half of the solution  $R^T \kappa$ **!** and the entire solution is thus

$$
R_{\kappa \ell}^{\sigma} = R_{\kappa \ell}^{\prime \sigma} + R_{\ell \kappa}^{\prime \sigma}
$$

We can determine  $C(m)$  from equation  $(A-3)$  by setting  $\gamma=0$  after differentiation.

$$
C(m) = \tanh(mh)
$$

The solution 
$$
R_{\kappa\ell}^{\prime}
$$
 is then  
\n
$$
R_{\kappa\ell}^{\prime\tau} = \int_{0}^{T} \left\{ \frac{\kappa - J}{2} \tanh\left(\frac{\kappa - J}{2} h\right) \left[ \hat{R}_{\kappa}^{\kappa} R_{\ell}^{\tau-k} \right] \int_{\ell, J-k} + \frac{\kappa - J}{2} \left[ \left( I + \tanh\left(\frac{\kappa - J}{2} h\right) \right) \tanh\left(\frac{J-k}{2} h\right) \right] \left[ R_{\kappa}^{\kappa} R_{\ell}^{\tau-k} \right] \int_{\ell, J-k+1}^{J-k+1} d\tau
$$

where the  $R_{\alpha}^{s}$  are functions of  $\tilde{\tau}$ . We substitute in for the **and integrate to obtain terms of the form:** and integrate to obtain terms of the form:

$$
R_{\kappa,\ell}^{'J} = \frac{\kappa - J}{2} \tanh\left(\frac{\kappa - J}{2}h\right) \int_{\ell, J-k} \chi(\kappa) \left\{ \frac{Q_{+}b_{-}}{\chi(\kappa) + \chi(J-\kappa-1)} \right\} \varepsilon
$$
  
+ 
$$
\frac{a_{+}b_{-}}{\chi(\kappa) - \chi(J-\kappa-1)} + \frac{a_{-}b_{+} \exp[i(\chi(\kappa) + \chi(J-\kappa-1))\varepsilon]}{\chi(\kappa) - \chi(J-\kappa-1)}
$$
  
+ 
$$
\frac{a_{+}b_{-}}{\chi(\kappa) - \chi(J-\kappa-1)} + \frac{a_{-}b_{+} \exp[i(\chi(\kappa) + \chi(J-\kappa-1))\varepsilon]}{\chi(\kappa) - \chi(J-\kappa-1)}
$$

$$
+\frac{a-b_{-}exp[z(\gamma(k)+\gamma(\tau-k-1)|t]}{\gamma(k)-\gamma(\tau-k-1)}\bigg\}+const.+other
$$
  
Terms in exp[ $\pm i(\gamma(k)+\gamma(\tau-k+1))t$ ]

To see what frequencies are present we look at the frequency of one term, e.g. the first term above. This term,  $\overline{L}$  is

$$
T_i = f_i \exp[i(Y(\kappa) - Y(\tau \kappa - 1))]t
$$

where  $\gamma(\kappa)$  is

$$
\gamma(\kappa) = \sqrt{\frac{\kappa - 1}{2} \frac{4}{3} \tanh\left(\frac{\kappa - 1}{2} \mu\right)}
$$

We assume that  $h$  is large (deep water) and we have

$$
\lambda(\kappa) = \pm \frac{5}{\kappa-1} \sqrt{3\mu}
$$

and

$$
\gamma(\tau_{-\kappa-1}) = \pm \frac{\kappa_{+1}}{2} \sqrt{4\mu}
$$

The frequencies present are then

$$
w_{\kappa} = \gamma(\kappa) - \gamma(\tau - \kappa - i) = \left(\pm \frac{\kappa - \tau}{2} \pm \frac{\kappa + i}{2}\right) \sqrt{9h}
$$

which are the sum and difference frequencies of all frequencies present. If we start with just a few frequencies we generate many more due to the nonlinearity of the problem. A study of the energy flow from one frequency to another is possible with the representation scheme used, but is quite tedious. We have shown here only part of the second order kernel,  $R_{\alpha\beta}^n$  which is itself quite cumbersome, and the higher order kernels are even worse. The only saving grace is that once the kernels are found the problem is solved for arbitrary initial conditions.

APPENDIX B

NORMALITY **TEST** FLOW GRAPH

Input **-** X(I) series, I=1,LX

Compute mean

$$
\chi_{\text{MEAN}} = \sum_{\mathbf{I}=\mathbf{I}}^{\mathbf{L}\mathbf{X}} \chi(\mathbf{I})/\mathbf{L}\mathbf{X}
$$

Compute standard deviation

$$
STDEFV = \left[\sum_{T=1}^{LX} (X(T) - X MEAN)^{2} / LX \right]^{1/2}
$$

Define NRANGE

**NRANGE =**  $\sqrt{LX}$ 

(This is an arbitrary definition. NRANGE should be small enough so that at least **5** values of X(I) fall in each range,) Find the X values which divide the normal density with mean **XMEAN** and standard deviation **STDEV** into NRANGE ranges of equal probability. Use SUBROUTINE NOINT2. Returns LRANGE(=NRANGE-1) values for range limits, RANGE(1).

First range is  $(-\infty, RANGE(1))$ , lst range is  $(RANGE(LRANGE), \infty)$ .

Count number of values falling in each range. Use SUBROUTINE FRQCT2. Returns fixed point count of number in each range in vector ICOUNT(I).

Chi Square test

P=1/NRANGE=probability of falling in any range.

$$
\chi^{2} = \sum_{\mathbf{T}=\mathbf{1}}^{NRANGE} (ICOUNT(\mathbf{I}) - P*K)^{2} / (P*K)
$$

Number of degrees of freedom=NRANGE-3. Use SUBROUTINE CHISQR.

Compute probability of exceeding  $\chi^2$ . Use SUBROUTINE KIINT1.

See APPENDIX G for program listings

APPENDIX C

EXPANSION OF EMPIRICAL PROBABILITY DENSITY FUNCTIONS ABOUT THE NORMAL DENSITY IN TERMS OF MOMENTS

It is possible to expand a probability density about the normal density if the moments higher than the mean and variance are known. It is not, however, guaranteed that the expansion will converge in all cases. If  $F(x)$  is the probability distribution, and  $f(x)$ ,

$$
f(x) = \frac{dF(x)}{dx}
$$

is the density and  $\varphi(x)$  is the normal density,

$$
\psi_{(x)} = \frac{1}{\sqrt{2\pi}} e^{-x^2/2}
$$

then the expansion in terms of the derivatives of the normal density, the Edgeworth series, is

$$
f(x) = C_0 \oint_{(x)} + \frac{c_1}{i!} \oint_{(x)}^{i} + \frac{c_2}{2!} \oint_{(x)}^{n} (x) + \dots
$$
 (c - 1)

and will converge if the integral

$$
\int_{-\infty}^{\infty} e^{-x^2/4} dF(x)
$$

converges and if  $f(x)$  is of bounded variation in  $(-\infty, \infty)$ (Cramer, 1946). For our purposes we need not worry too much about the convergence. We only wish to see if we can approximate the distribution fairly well with just a few terms of the expansion.

It is now possible to obtain the coefficient  $C_n$  in terms of the moments. Remembering that the normal density,  $\mathcal{X}^{(\chi)}$  is the "generating function" for Hermite polynomials

$$
\left(\frac{d}{dx}\right)^n e^{-x^2/2} = (-1)^n H_n(x) e^{-x^2/2}
$$
 (c-2)

where  $H_n(x)$  is the nth order Hermite polynomial, and that the Hermite polynomials are orthagonal with respect to  $\varphi(x)$ 

$$
\int_{-\infty}^{\infty} H_m(x) H_n(x) \psi_{(x)} dx = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} H_m(x) H_n(x) e^{-x^2/2} dx
$$

$$
= \begin{cases} n! & \text{for } m=n \\ 0 & \text{for } m \neq n \end{cases} (c-s)
$$

we can now solve for the  $C_n$ Substituting  $\psi_{(x)}^{(n)} = (-1)^n H_n(x) \psi(x)$  into equation (C-1) we have

$$
\int (x) = C_0 \text{H}_0(x) + C_1 \frac{(-1)}{1!} H_1(x) + C_2 \frac{(-1)^2}{2!} H_2(x) + ... + \frac{C_n(-1)^n}{n} H_n(x) + C_{n-4}
$$

Multiplying both sides by  $H_m(x)$  and integrating we have, because of **(C-3),**

$$
C_m = (-1)^m \int_{-\infty}^{\infty} H_m(x) f(x) dx
$$
 (C-5)

Since  $H_{m}(\chi)$  is a polynomial and  $f(\chi)$  is a probability density

the integral is simply a sum of moments. The moments (central moments) are  $M_{\mathbf{k}}$  where

$$
M_{\kappa} = \int_{-\infty}^{\infty} (\zeta - m)^{\kappa} f(\zeta) d\zeta
$$
 (c-6)

and  $w$  is the mean. The unit normal density (zero mean, unit standard deviation) was assumed in this derivation so that  $f(x)$  must be the function of the standardized variable  $\frac{\zeta - m}{\zeta}$  where  $\int$  is the standard deviation. This means that the  $\zeta$ -<sup>+</sup>h moment of the standardized variable is  $\frac{1}{\sqrt{2}}$  **.** Hence  $C_0 = 1$  **,**  $C_1 = C_2$  $H_3(x) = X^3 - 3X$ , and so from (C-5),  $C_3 = -\frac{33}{3}$  The rest of the  $C_n$  may be obtained from the  $H_n(x)$  in the same manner. Thus

$$
C_{4} = \frac{u_{4}}{\sigma^{4}} - 3
$$
  

$$
C_{5} = -\frac{u_{5}}{\sigma^{5}} + 10 \frac{u_{3}}{\sigma^{3}}
$$
  

$$
C_{6} = \frac{u_{6}}{\sigma^{6}} - 15 \frac{u_{4}}{\sigma^{4}} + 30
$$

The moments may be estimated from the data **by** averaging so that the integral **(A-6)** need not be performed.

The computation of the approximations using up to  $C_{\boldsymbol{\omega}}$  has been programmed **by** Roy Greenfield. (See SUBROUTINE PRBFIT in APPENDIX **G.)** The expressions for the approximations which must be evaluated are

$$
f_{1}(x) = \left[1 + \frac{\mu_{3}}{6 \sigma_{3}}(x^{3} - 3x)\right] \varphi_{(x)}
$$
  
\n
$$
f_{2}(x) = f_{1}(x) + \left[\left(\frac{\mu_{4}}{24\sigma_{4}} + \frac{1}{8}\right)(x^{4} - 6x^{2} + 3)\right] \varphi_{(x)}
$$
  
\n
$$
f_{n}(x) = f_{n-1}(x) + \left[\frac{c_{n}}{n!}(-1)^{n} H_{n}(x)\right] \varphi_{(x)}
$$

Care must be taken that the  $\chi^1$   $\mathsf S$  are the values of the standardized  $variables.$ 

## APPENDIX D

**INDEPENDENCE AND DEPENDENCE MEASURES**

### Poker Count Test for Independence

Given a series of equally likely integers from zero to nine it is possible, under the assumption that the numbers are independent, to compute the probable number of non-overlapping groups of five numbers which fall into each of eight categories. These categories are similar to those of a poker game where each group of five is considered a hand and each hand has a certain value. The analogy to the poker game is not completely accurate since the "card" values are 0 to **9** rather than ace to king, and it is possible to have five of a kind. Also the series, which takes the place of the card deck, has many more than 52 numbers in it, and removal of a number does not decrease its later probability of occurrence. The eight categories or hand types with their respective probabilities are (Durand, 1962, personal communication):



These probabilities are exact. The decimals terminate at the fourth place. In assigning a hand to one of the categories the order of the digits within the group of five does not matter.

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If the series of numbers is independent, then it is expected that the number of each type of hand will be approximately the probability for that hand times the total number of hands. Both this test and the mean square contingency test require a mapping of the given series into an integer series. The poker count test requires that the ten digits have equal probability. Hence the probability density of the original series is transformed into a rectangular density and the original series is mapped into an integer series with values from zero to nine with each integer having probability .1. Figure D-4 shows the steps necessary in the poker count test and APPENDIX G contains program listings.

Transformation of Probability Densities

Suppose  $P_{\varphi}(x) = f(x)$  is the probability density (frequency function) of a random variable  $\begin{bmatrix} \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot \end{bmatrix}$  . The distribution function is then

$$
Q(x) = \int_{-\infty}^{x} f(y) \, dy = F(x)
$$

The change of variable,  $y = F(x)$  is known as the "probability" transformation" (Wadsworth and Bryan, 1960).

The probability density  $P_{\gamma}(y)$  can be found as follows:

$$
P_{\eta}(y) dy = P_{\xi}(x) dx
$$
  

$$
P_{\eta}(y) = P_{\xi}(x) \frac{dx}{dy} = \frac{f(x)}{f(x)} = 1
$$

The variable  $\zeta$  is thus rectangularly distributed and, since  $F(x)$ is defined from 0 to 1,  $0 \le y \le l$ .

For the joint distribution,  $P_{\{1, \{2\}}(X_1, X_2)$ , using the same transformation, we have

$$
P_{\zeta_1 \zeta_2}(x_1, x_2) = P_{\zeta_1}(x_1) P_{\zeta_2 | \zeta_1}(x_1, x_2)
$$

where  $P_{\{i\}_{\{1,\}}(x_i|x_i)d x_i d x_2$  denotes the compound probability that  $x_2 < \zeta_2 < x_2 + dx_2$  given that  $x_i \le \zeta_i \le x_i + dx_i$ . Using the same transformation,  $y = F(x)$ , we have

$$
P_{\gamma_1, \gamma_2}(y_1, y_2) dy_1 dy_2 = P_{\xi_1}(x_1) P_{\xi_2 | \xi_1}(x_2 | x_1) dx_1 dx_2
$$

The Jacobian for this transformation,  $J$ , gives

$$
J = \begin{bmatrix} dx, dx_1 = 1 \text{ J} d_1, dy_2 \\ \frac{\partial x_1}{\partial y_1} & \frac{\partial x_2}{\partial y_2} \\ \frac{\partial x_3}{\partial y_1} & \frac{\partial x_3}{\partial y_2} \end{bmatrix} = \begin{bmatrix} \frac{1}{f(x)} & 0 \\ 0 & \frac{1}{f(x)} \end{bmatrix}
$$
  

$$
J = \left[ \frac{1}{f(x)} \right]^2
$$
  

$$
J = \left[ \frac{1}{f(x)} \right]^2
$$
  

$$
P_{\eta, \eta_2}(y_1, y_2) = \frac{P_{\eta, (x_1)} P_{\eta, \eta_2}(x_2 | x_1)}{\left[ f(x) \right]^2}
$$
  

$$
P_{\eta, \eta_2}(y_1, y_2) = \frac{P_{\eta, \eta_2}(x_1 | x_1)}{f(x_2 | x_1)}
$$

If 
$$
\int_{1}^{x} \text{ and } \int_{2}^{x} \text{ are independent then}
$$
  

$$
\int_{1}^{x} |f(x)| dx = \int_{1}^{x} (x_1) \cdot f(x) \text{ and}
$$

$$
\int_{1}^{x} \int_{1}^{x} (y_1, y_2) = 1
$$

The result is that if  $\zeta$  and '  $\zeta$ <sub>2</sub> are independent, then  $\gamma$ , and  $\gamma_1$ are also independent, and if  $\zeta_1$  and  $\zeta_2$  are dependent, then  $\gamma$ , and  $\gamma_1$ are also dependent. The compound probabilities will differ **by** a factor equal to  $\left| \frac{1}{f(x)} \right|$ 

$$
P_{\eta_{2}}|\eta_{1}(y_{2}|y_{1}) = P_{\eta_{2}}|_{\eta_{1}}(x_{2}|x_{1}) \left| \frac{1}{f(x)} \right|
$$

 $\mathbf{I}$ 

If  $\int$ , and  $\int_2$  are independent, then all of the higher probability densities for  $\gamma$  are rectangular. An extension of this can easily be made for any number of random variables, and in particular for five variables as is necessary for the poker count test.

Mean Square Contingency and Dependency Measure

The measure of the degree of dependence of two variables which has been used is related to the mean square contingency (Cramer, **1951).**

Suppose that two variables,  $\zeta$  and  $\gamma$  have densities  $P_{\zeta}(x_1)$ and  $P_{\gamma_1}(\gamma_i)$  and a joint density  $P_{\gamma_1}(X_i, \gamma_i)$  where  $X_i$  and  $Y_j$  are discrete and **L** : , *)N I). ,*

Hence

$$
\sum_{i} P_{\xi} \gamma(x_i, y_j) = P_{\gamma}(y_i)
$$

$$
\sum_{j} P_{\xi} \gamma(x_i, y_j) = P_{\xi}(x_i)
$$

The mean square contingency,  $\varphi$ <sup>2</sup> is defined as

$$
\varphi^{2} = \sum_{i} \sum_{j} \frac{(P_{\xi} \eta(x_{i}, y_{j}) - P_{\xi}(x_{i}) P_{\eta}(y_{j}))^{2}}{P_{\xi}(x_{i}) P_{\eta}(y_{j})}
$$

$$
= \sum_{i} \sum_{j} \frac{\left[ P_{\xi \gamma}(x_{i}, y_{j}) \right]^{2}}{P_{\xi}(x_{i}) P_{\gamma}(y_{j})} - 1
$$

If and only if the variables are independent

$$
P_{\overline{y}}\gamma\left(\overline{x_i},\overline{y_j}\right) = P_{\overline{y}}(x_i) P_{\overline{y}}(y_j)
$$
and  $\varphi^2 = 0$ .

Since

$$
P_{\xi \gamma}(x_i, y_i) = P_{\xi}(x_i) P_{\gamma | \xi}(y_i | x_i) = P_{\gamma}(y_i) P_{\xi | \gamma}(x_i | y_i)
$$

and all probabilities are less than or equal to one,

$$
P_{\xi\gamma}(x_i, y_j) \leq \begin{cases} P_{\gamma}(y_i) \\ P_{\xi}(x_i) \end{cases}
$$

thus

$$
\sum_{i,j}^{M,m} \frac{P_{\gamma}^{2} \gamma(x_{i},y_{j})}{P_{\gamma}(x_{i}) P_{\gamma}(y_{j})} \leq \varphi
$$

and

$$
\varphi^2 \leq \varphi - I
$$

where  $\mathbf{Q}$  is the smaller of  $\mathbf{N}$  and  $\mathbf{M}$ , the limits of the sumation. Therefore the quantity  $\int_{0}^{3}$  ( $\int_{0}^{3}$ -1), which we will call the dependency, may be used as a standard measure of dependence since

$$
0\leq \frac{\varphi^2}{\varphi-1}\leq 1
$$

There is, of course, some difficulty in using this or any measured dependence on numerical data. Numbers generated **by** independent random processes will not in general give a zero value for the dependency. The question arises, therefore, as to the interpretation of the number resulting from the dependency test. Since it is uncertain how large the dependency can be and the series still remain independent, a number of tests were run on independent random numbers. The numbers were obtained from the Rand Corporation on punched cards and are the same as the numbers which appear in the book, **1,000,000** Random Digits (Rand Corporation **1958).** These numbers were generated **by** an independent process.

The numbers were run through both the poker count test and the dependency test. Three different lengths of series were used, **3000, 2500** and 2000, and each was repeated **8** times so that a mean and cariance could be computed. The results of the dependency test are shown in Figure **D-1.** Straight lines have been dotted in to indicate the mean and standard deviation changes with series length. There is no reason to suspect that

their values actually fall on a straight line, in fact one would suspect that the line would curve off concave upward on the right and concave downward to the left. These tests were carried out for a lag of one, that is the random variables took on values of  $\chi_n$  and  $\chi_{n+1}$  of the series of digits.

Since it is important that the denominator not be zero, the series of real data were mapped into integer series from 1 to 10 with rectangular densities. This was, of course, not necessary with the Rand random digits, since they were already equally likely integers. However, one was added to each Rand digit so that the series would be from 1 to 10 rather than 0 to 9. This was necessary only for ease and speed of computation of the second probability density. Figures **D-2** to D-5 show flow graphs of the steps necessary to compute the empirical probability density and perform the probability transformation, the poker count test and the mean square contingency test. APPENDIX G contains the listings of the programs used in these operations.



Empirical Probability Density Flow Graph

```
Inputs - X(I) series, I=1, LX
```
NDIV number of ranges

Find maximum, XMAX, and minimum, XMIN, of X series.

Compute range limits for NDIV equally spaced ranges from

XMIN to XMAX

RANGE(I)=XMIN+(I-1) (XMAX-XMIN)/NDIV, I=1, NDIV+1

NDIV is somewhat arbitrary. It should be much smaller than LX, the length of the X series. We have used NDIV=100 with LX 2500.

Count number of values of X(I) falling in each of the NDIV ranges. Use SUBROUTINE FROCT2.

NOTE **-** FRQCT2 assumes that the NDIV+1 range limits define NDIV+2 ranges. The count vector, ICOUNT(I), I=1,NDIV+2, must therefore be altered such that ICOUNT(2)=ICOUNT(2)+ICOUNT(1), and ICOUNT(NDIV+1)=ICOUNT(NDIV+1)+ICOUNT(NDIV+2). The correct counts are then in ICOUNT(2) to ICOUNT(NDIV+1). This may then be normalized to give the frequency ratio or probability density, PROB(I).

 $PROB(I)=ICOUNT(I) NDIV/(LX (XMAX-XML))$ 

Figure D-2

Probability Transformation Flow Graph

Rectangularize Probability Density

Inputs **-** PROB(I), I=1,NDIV, The probability density normalized such that

$$
\sum_{x=1}^{N\text{Div}} PROB(I) \Delta x = 1 \text{ j} \quad \Delta x = (x \text{max} - x \text{min})/Lx
$$

XMIN **=** Minimum value of original time series

XMAX **=** Maximum value of original time series

NPROB =Number of ranges of equal probability desired.

Need not equal NDIV

X(I),I=1,LX, the time series

Find X limits which divide the empirical density into NPROB ranges of equal probability, XLIMIT(I), I=1,NPROB+1.

(Linear interpolation where necessary) Use SUBROUTINE GRUP2

Map X(I) series into IX(I) series (integer series such that for **XLIMIT(J) X(I) XLIMIT(J+1), IX(I)=J-1+IXLO**

where IXLO can be adjusted to give desired **d.c.** level.

Use **SUBROUTINE MPSEQ1**

Result is interger series IX(I), I=1,LX with NPROB different values from IXLO to IXLO+NPROB-1 with equal probability, 1/NPROB Poker Count Test Flow Graph

Inputs  $- X(I)$ , I=1, LX time series

LX length of series

Compute empirical probability density. See Figure D-2 for flow graph of this procedure

Perform probability density transformation to map X(I) series into IX(I) series with

# $0 \leq IX(1) \leq 9$

See Figure D-3 for flow graph of this procedure with IXLO=0. Take IX(I) series  $\ln$  non-overlapping groups of  $5, IX(I)$ , I=1, ...  $5, IX(I), I=6, \ldots.10,$  etc and consider these as poker hands. Evaluate the poker hands and count number of each type. (Types - bust, 1 pair, 2 pair, 3 of a kind etc.) Total number of hands is LX/5 rounded down. USE SUBROUTINE POKCT1. Compare with theoretical count for independent series.

(See a priori probabilities on first page of this APPENDIX.)

Figure D-4

Mean Square Contingency and Dependency Test Flow Graph

Inputs **-** X(I), I=1,LX time series

LX length of series

Compute empirical probability density. See Figure **D-2** for flow graph of this procedure.

Perform probability density transformation to map X(I) series into

IX(I) series with  $1 \leq IX(1) \leq JHIGH$ , where JHIGH  $\leq 25$ .

(Requirement of **SUBROUTINE** PROB2 used below.)

Note - If poker count test is also done the mapped series used there can be used here if one is added to every IX value. JIIGH will be **10** for this case.

(See Figure **D-3** for transformation and mapping flow graph.) Compute second probability density, **P(I,J)** for lag of one.

Use **SUBROUTINE** PROB2. (Gives joint probability that IX(I)=L

and  $IX(I+1)=M$  for  $I=1$ ,  $LX-1$ , and M and  $L \geq 1$ ,  $\leq JHIGH$ .)

Compute mean square contingency and dependency.

$$
M.S.C. = \sum_{\mathbf{I}=\mathbf{I}} \sum_{\mathbf{J}=\mathbf{I}}^{T} \left[ \left( P(\mathbf{I},\mathbf{I}) \right)^2 / \left( P(\mathbf{I}) \cdot P(\mathbf{I}) \right) \right] - 1
$$

where

$$
P(I) = \sum_{\tau=1}^{T+IGH} P(L,T) \neq 0 \ , \ P(J) = \sum_{\tau=1}^{T+IGH} P(T,T) \neq 0
$$

DEPENDENCY=M.S.C./ (JHIGH-1)

**USE** SUBROUTINE **MSCON1.**

Figure **D-5**

### APPENDIX E

#### FACTORIZATION OF THE POWER SPECTRUM

The problem of spectrum factorization in the frequency domain was solved by Kolmogorov (1941). The treatment here is similar to Robinson (1956).

Given a power density spectrum,  $\oint_{\mathcal{L}}(\omega)$ , it is possible to factor it such that

$$
\Phi(\omega) = B(\omega) \overline{B(\omega)}
$$

where

$$
B(w) = \sqrt{\Phi(w)} e^{\lambda \Theta(w)}
$$

That this factorization is possible is quite obvious and, in fact, an infinite number of such factorization exist. The trivial case is  $\Theta(\omega)$  = 0. There is, however, one important case, and that is when  $\Theta(\omega)$ has no poles or zeros in the lower half of the  $\lambda$  plane  $(\lambda : \mathsf{W} + \mathsf{L}^T)$ (Lee, **1960).** In this case **B3L** W) corresponds to the transfer function of a physically realizable system, that is, a system which does not have output before it has input. **A** pole in the lower half of the **X** plane transforms to the negative time axis and can therefore be considered a "source" for negative time. If  $\beta(\omega)$  has poles in the lower half plane, its Fourier transform  $\mathcal{B}(t)$  will only be non-zero for  $t \geq 0$ , and  $\beta(t)$  then said to be one-sided in positive time. If  $\beta(u)$ also has no zeros in the lower half plane, then its inverse  $\sqrt{\mathsf{B}(\omega)}$  will have no poles in the lower half plane and its Fourier transform will also be one-sided.  $B(t)$  is then called the minimum phase wavelet. The factorization problem is the problem of finding  $\beta$ (+) from  $\oint(u)$  and can be solved as follows.

If we take the  $\overline{z}$  transform, i.e.  $\overline{z}$  =  $\overline{e}^{i\omega}$ , of  $B(\omega)$  to obtain **Z1)** , we have mapped the lower half of the place into the interior of the unit circle and we now consider  $\mathbb{B}(\mathbf{r})$  a polynomial in  $\mathbf{r}$ . That is  $\mathcal{G}(\omega)$  is the Fourier transform of some time function  $\mathcal{G}(t)$  and as such has the form **0)**

$$
B(\omega) = \sum_{s=-\infty}^{\infty} b_s e^{-i\omega s}
$$

and the **2** transform becomes

$$
B(\mathbf{z}) = \sum_{s=-\infty}^{\infty} b_s z^s
$$

and **3** must have no poles or zeros inside the unit circle, There are certain restrictions on  $\Phi$ <sub>( $\omega$ )</sub>, namely

1. 
$$
\oint(w) = 0
$$
  
\n2.  $\int_{-\pi}^{\pi} \log \overline{\oint}(w) dw > -\infty$   
\n3.  $\int_{-\pi}^{\pi} \overline{\oint}(w) dw < \infty$ 

which must be met if  $\mathbf{B}(\mathbf{z})$  is to exist. If condition (1) is not met, then the integral (2) will not converge. Condition (2) is equivalent to the Paley-Wiener criterion (Robinson, 1954, **p.** 149) and is a requirement for the existence of a moving average and an autoregressive representation of the time series. Condition **(3)** states that the power must be finite and is just a stability requirement.

If these requirements are fulfilled, then the logarithm of  $\beta(\xi)$  will be analytic for  $|2 \n\leq l$ .

$$
\log B(\omega) = \frac{1}{2} \log \Phi(\omega) + i \Theta(\omega)
$$
  
or  

$$
\log B(\omega) = u(\omega) + i \mathbf{V}(\omega)
$$

Hence the problem of obtaining the minimum phase wavelet is now one of finding the imaginary part,  $\nabla(\blacktriangleright)$ , of a function analytic inside the unit circle given the real part,  $U(\tilde{\tau})$ , on the circle. This is also the potential theory problem of finding the field inside of a region given the sources on the boundary. The function  $\log \beta(2)$  can be expressed as a power (Taylor) series in its region of analyticity

$$
\log B(x) = \sum_{r=-\infty}^{\infty} a_r z^r
$$

Expanding log  $B(z)$ : log  $B(r e^{i\omega})$  in a Fourier series  $log B(re^{iw}) = U(re^{iw}) + i v(re^{iw})$  $=\sum_{k} k_{k} e^{i\omega k}$ ,  $x_{k} = c_{k} + id_{k}$  $U(re^{iw}) = Re\left[\sum_{C_{\kappa}+id_{\kappa}}r^{\kappa}e^{iw_{\kappa}}\right]$  $R$ e  $\Sigma$ c<sub>k</sub>cos  $\kappa w$ + $id$ <sub>k</sub>cos  $\kappa w$ + $ic$ ks $in \kappa w$ - $d_{\kappa}$ s $in \kappa w$ <sup>k</sup>  $= \sum_{k=0}^{n} (C_k \cos \kappa w - d_k \sin \kappa w) r^k$ 

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However

$$
u(re^{i\omega})=\frac{1}{2}log\ \Phi(w)
$$
 at r=1

and  $\overline{\Phi}(\omega)$  is an even function, i.e.  $\Phi(\omega) = \Phi(-\omega)$ 

since

$$
\Phi(\omega) = \sum_{s} \phi_{s}^{s} \cos \omega s
$$

Therefore  $1/2 \log \Phi(w)$  is also even

 $x_K = c_K$ 

 $d_{\kappa} = 0$ and

Hence

$$
\frac{1}{2} \log \Phi(w) = \sum \alpha \kappa \cos \kappa w
$$
  

$$
\alpha \kappa = \frac{1}{\pi} \int_{-\pi}^{\pi} \frac{1}{2} \log \Phi(w) \cos \kappa w dw
$$

and

and

The wavelet  $b_s$  is then determined from

$$
\mathbf{B}(\mathbf{z}) = \sum_{s=0}^{\infty} b_s \mathbf{z}^s = \exp\left[\sum_{k=-\infty}^{\infty} \alpha_k \mathbf{z}^k\right] = \exp\left[\sum_{k=-\infty}^{\infty} \frac{1}{4\pi} \int_0^{\pi} \log \Phi(\omega) \right]
$$
\n
$$
\cos \kappa \omega d\omega \mathbf{z}^k
$$

The following method, suitable for programming purposes, for getting the bs was first given in MIT G.A.G. Report **9 (1956)** and was repeated in Simpson et al (1962a).

The **bs** will have to be cut off after some **S** value, say  $S =$ It is shown below that the first points in the wavelet) may be obtained exactly from the first  $M + | \mathcal{X}|$  $m + 1$  terms of  $b$ s (the first  $m + 1$ 

$$
\sum_{\substack{S=0}}^{\infty} b_{s} Z^{S} = e^{M_{0}} \left[ 1 + \frac{2 M_{1}}{1!} \frac{2}{2} + \left( \frac{2 M_{1}}{2!} \right)^{2} \frac{2}{2} \frac{2}{4} ... \right] \left[ 1 + \frac{2 M_{2}}{1!} \frac{2}{2} + \left( \frac{2 M_{3}}{2!} \right)^{2} \frac{2}{4} ... \right]
$$
  

$$
X \left[ 1 + \frac{2 M_{3}}{1!} \frac{2}{2} + \left( \frac{2 M_{3}}{2!} \right)^{2} \frac{2}{2} + ... \right] \left[ ... \right] ...
$$
  

$$
X \left[ 1 + \frac{2 M_{m}}{1!} \frac{2}{2} + \left( \frac{2 M_{m}}{2!} \right)^{2} \frac{2}{2} \frac{2}{4} ... \right] \left[ ... \right] ...
$$

Matching like powers of  $\mathcal{Z}$  we find

 $b_0 = e^{a_0}$  $b_1 = e^{\alpha \cdot (2 \cdot x)}$  $b_2 = e^{\alpha_0}\left[ \left( \frac{2\alpha_1}{2!} \right)^2 + \frac{2\alpha_2}{1!} \right]$ 

In general, if we are interested in obtaining  $b_0, \ldots, b_m$ , we may drop terms in any polynomial with exponents  $\mathcal{V}$  and we may drop all polynomial twhose first power of  $\geq$  is  $\geq$  M . We also do not care about any cross terms whose  $\geq$  exponents are  $\geq$   $\mathsf{M}$  .

We disregard  $e^{\mathbf{d}_0}$  for the time being and consider the problem as follows:

 $\sum_{s=0}^{m} b_s$   $\overline{z}^s$  = (First  $m+1$  terms of)  $P_1(\overline{z}) P_2(\overline{z}) ... P_m(\overline{z})$ (this is just another way of grouping the terms).

Where 
$$
P_i = 1 + C_{i1} Z + C_{i2} Z^2 + ... + C_{i} m Z^m
$$

and

$$
C_{ij} = \left\{\begin{matrix}\left[\left(\frac{2\alpha_i}{i}\right)\left(\frac{2\alpha_i}{2}\right)\left(\frac{2\alpha_i}{3}\right) \cdots \left(\frac{2\alpha_i}{j/2}\right) & \text{for } j = ki\right] \\
0 & \text{for } j \neq ki\end{matrix}\right\}
$$

 $C_i$   $o = 1$ 

K is a positive integer. Considering  $b_s$  and  $c_{is}$  $\overline{\mathbf{a}}$ s time functions we may now consider the problem as one of partial convolution. Let  $\overline{F}$  stand for "First  $m+1$  terms of." Then

$$
P = \mathsf{E}(c \cdot \ast c^3 \ast c^3 \cdots \ast c^m)
$$

and

$$
b = F(c_1 * F(c_2 * F(c_3 * ... F(c_{m-1} * c_m)))...)
$$

Let 
$$
b^{(m)} = C_m
$$
  
\n
$$
b^{(m-1)} = F(C_{m-1} * C_m) = F(C_{m-1} * b^{(m)})
$$
\n
$$
b^{m-2} = F(C_{m-2} * F(C_{m-1} * C_m)) = F(C_{m-2} * b^{m-1})
$$
\n
$$
b^{(1)} = F(C_1 * b^2) = b
$$

Examination shows that  $b^{(l+1)}$  may be obtained from  $b^{(l)}$  by

the following formula representing partial convolution

$$
b_{s}^{(\ell-1)} = \sum_{i=0}^{s} C_{\ell-1, s-1} b_{i}^{(\ell)}
$$
  
5 = 0, 1, 2, ..., m

Further examination shows that  $\mathbf{b}^{n}$ , where  $\mathbf{M} = \mathbf{I} + \text{integral part}$ 

of  $M/2$ , may be written down by inspection

$$
b_{m}^{(m)} = 0
$$
\n
$$
b_{1}^{(m)} = 0
$$
\n
$$
b_{2}^{(m)} = 0
$$
\n
$$
\vdots
$$
\n
$$
b_{m}^{(m)} = C_{m,m}
$$
\n
$$
b_{m+1}^{(m)} = C_{m+m}
$$
\n
$$
\vdots
$$
\n
$$
C_{m,m}
$$

This can be seen by noting first that  $O_0^{\bullet\bullet}$   $\stackrel{\bullet}{\bullet}$  for all  $\Box$ and  $b_s^2$  = 0 for  $1\angle S\angle L$  and that the C<sub>LS</sub> for  $M_2 \leq L_2$  **in** have only two terms in them. As the partial convolution proceeds, the  $\mathbf{b}_{\rho}$  terms pickup the diagonal terms in the  $\mathbf{c}_{ij}$  matrix, and there are no other contributions to the next  $b_5^{(1)}$  until  $\angle$  2  $m/2$ . It can be seen that only one column of the  $C_{i,j}$  matrix is needed at a

time.

**A** program has been written for the spectrum factorization problem for 709 or 7090 computers. The program makes sure that  $\Phi(\omega) > 0$  by setting any value of  $\Phi(\omega)$  which is less than 10<sup>-6</sup> of the maximum value of  $\overrightarrow{\mathbf{\Phi}}(\mathbf{w})$  equal to 10<sup>-6</sup> of the maximum. The Daniell method of spectral estimation guarantees  $\Phi$ ( $\omega$ )> 0 but other spectral window such as the Turkey-Hamming window do not have the guarantee. The computation of the  $\alpha'$ 'S in the computation of the cosine expansion of  $\frac{1}{2}$   $\ell_{og}$   $\Phi(w)$ was done **by** trigonometric interpolation (Lanczos, **1956)** so that the integral need not be computed. The program FACTOR is listed in APPENDIX **G.** APPENDIX F

CONSTRUCTION OF THREE WHITE LIGHT SERIES WITH SPECIFIED **COHERENCES**

We wish to construct three unit variance white light series  $X_t^l$ ,  $X_t^2$ ,  $X_t^3$  with controlled coherences

$$
Coh_{12}(\omega) = \frac{\left|\oint_{12}(\omega)\right|}{\sqrt{\oint_{11}(\omega)\oint_{22}(\omega)}} = \sqrt{x_{12}(\omega)}
$$

Coh<sub>3</sub>(
$$
\omega
$$
) =  $\frac{|\Phi_{13}(\omega)|}{\sqrt{\Phi_{11}(\omega) \Phi_{33}(\omega)}}$  =  $\alpha_{13}(\omega)$  (F-1)

$$
Coh_{23}(\omega) = \frac{|\Phi_{23}(\omega)|}{\sqrt{\Phi_{22}(\omega) \Phi_{33}(\omega)}} = \alpha_{23}(\omega)
$$

The solution is an obvious extension of the Simpson et al **(1962)** treatment of constructing a pair of series with controlled coherence. Since  $X_t^1$ ,  $X_t^2$ ,  $Y_t^3$  are unit variance white light their spectra are

$$
\Phi_{11}(\omega) = \Phi_{22}(\omega) = \Phi_{33}(\omega) = \frac{1}{2\pi}
$$

 $\hbox{\bf hence}$ 

$$
\left| \Phi_{ij}(\omega) \right| = \frac{\alpha_{ij}(\omega)}{2\pi} , \quad i \leq j \leq 3
$$

or for zero phase shift

$$
\Phi_{ij}(\omega) = \frac{\alpha_{ij}}{2\pi}
$$

We assume that  $\chi_t^1$ ,  $\chi_t^2$  and  $\chi_t^3$  are broken up to have common and uncorrelated parts

$$
\chi_{t}^{1} = \chi_{t}^{c_{1}} + \chi_{t}^{c_{3}} + \chi_{t}^{R_{t}} + \chi_{t}^{R_{t}}
$$
\n
$$
\chi_{t}^{2} = \chi_{t}^{c_{1}} + \chi_{t}^{c_{2}} + \chi_{t}^{R_{2}}
$$
\n
$$
\chi_{t}^{3} = \chi_{t}^{c_{1}} + \chi_{t}^{c_{2}} + \chi_{t}^{C_{3}} + \chi_{t}^{R_{3}}
$$
\n(F-2)

 $\sim$ 

where all cross correlations

$$
\varphi_{c_i c_j}, \varphi_{R_i R_j}, \varphi_{i \neq j}
$$
\n
$$
\varphi_{c_i R_j}, \varphi_{i \neq 1, 2, 3}, \varphi_{j \neq 1, 2, 3}
$$

 $\sim$   $\epsilon$ 

are zero. The autospectra of the  $\chi_t^{\mu}$  series are then

$$
\begin{aligned}\n\oint_{\mathfrak{h}} (\omega) &= \bar{\Phi}_{c_1}(\omega) + \bar{\Phi}_{c_2}(\omega) + \bar{\Phi}_{R_1}(\omega) = \frac{1}{2\pi} \\
\oint_{22} (\omega) &= \bar{\Phi}_{c_1}(\omega) + \bar{\Phi}_{c_2}(\omega) + \bar{\Phi}_{R_2}(\omega) = \frac{1}{2\pi} \\
\oint_{33} (\omega) &= \bar{\Phi}_{c_1}(\omega) + \bar{\Phi}_{c_2}(\omega) + \bar{\Phi}_{c_3}(\omega) + \bar{\Phi}_{R_3}(\omega) = \frac{1}{2\pi}\n\end{aligned}
$$

The cross-spectra are

$$
\overline{\Phi}_{12}(\omega) = \overline{\Phi}_{c_1}(\omega) = \frac{\alpha_{12}(\omega)}{2\pi}
$$
\n
$$
\overline{\Phi}_{13}(\omega) = \overline{\Phi}_{c_1}(\omega) + \overline{\Phi}_{c_3}(\omega) = \frac{\alpha_{13}(\omega)}{2\pi}
$$
\n
$$
\overline{\Phi}_{23}(\omega) = \overline{\Phi}_{c_1}(\omega) + \overline{\Phi}_{c_2}(\omega) = \frac{\alpha_{23}(\omega)}{2\pi}
$$

We therefore have

215

$$
\Phi_{c_{2}}(\omega) = \frac{\alpha_{23}(\omega) - \alpha_{12}(\omega)}{2\pi}
$$
\n
$$
\Phi_{c_{3}}(\omega) = \frac{\alpha_{13}(\omega) - \alpha_{12}(\omega)}{2\pi}
$$
\n
$$
\Phi_{R_{1}}(\omega) = \frac{1 - \alpha_{13}(\omega)}{2\pi}
$$
\n
$$
\Phi_{R_{2}}(\omega) = \frac{1 - \alpha_{23}(\omega)}{2\pi}
$$
\n
$$
\Phi_{R_{3}}(\omega) = \frac{1 + \alpha_{12}(\omega) - \alpha_{23}(\omega) - \alpha_{13}(\omega)}{2\pi}
$$

We must first construct the six mutually independent series  $X_t^{c_i}$ ,  $X_t^{R_i}$ , i=1,2,3 with the power spectra  $\Phi_{c_i}$ ,  $\Phi_{R_i}$  given above. We then con-<br>struct the  $X_t$  series with equations F-2. These series have the coherences  $\alpha_{ij}(\omega)$  as shown in equations F-1.
#### APPENDIX G

#### PROGRAM LISTINGS

Listings, with descriptions and examples, of some of the more impor- .tant programs used in the computations in this thesis. The listings are in alphabetical order and include all subroutines appearing in the transfer vectors with the exception of the FORTRAN System routines. An index of these programs and other programs useful in time series analysis appears in Scientific Report Number 4 of Contract AF 19(604)7378 (Simpson et al, 1962b) and complete listings will appear (Simpson, 1963, in press) in book form in the near future. All the programs appearing here are designed to operate under the FORTRAN-II system for the IBM 709-7090 computers.

Throughout the listings the terms FORTRAN INTEGER, FORTRAN II INTEGER, and INTEGER are synonomous and refer to a fixed point integer in the decrement. The terms MACHINE LANGUAGE INTEGER, MACHINE INTEGER and MLI refer to a fixed point integer in the decrement. The terms LSTHN and LSTHN = are equivalent to  $\leq$  and  $\leq$  while GRTHN and GRTHN = are equivalent to > amd **>.** It should be noted that expressions which appear in the "ABSTRACT" section of the writeup may deviate from the usual FORTRAN conventions.

```
21S
```




 $220$ 

COSP		PROGRAM LISTINGS	<b>COSP</b>	
***********************			**************	
٠ FAP	COSP (SUBROUTINE)		2/18/63 LAST CARD IN DECK IS NO. 0844	0001
∗COSP				0002
<b>CCUNT</b>	-1000			0003
LBL	<b>CCSP</b>			0004
<b>ENTRY</b>		CCSP (SSX,ASX,L,COSTAB,M,JMIN,JMAX,TYPE,COSTR)		0005
		ENTRY SISP (SAX, AAX, L, SINTAB, M, JMIN, JMAX, TYPE, SINTR)		0006
ENTRY			CCSISP (SSX, ASX, SAX, AAX, L, COSTAB, SINTAB, M, JMIN, JMAX, TYPE,	<b>COO7</b>
۰	CCSTR, SINTR)			0008
				0009
	----ABSTRACT----			0010 0011
		TITLE - COSP WITH SECONDARY ENTRY POINTS SISP AND COSISP		0012
٠			FAST COSINE AND/OR SINE TRANSFORMS FROM 2 OR 4 EVEN-ODD PARTS	0013
۰				0014
٠			COSP COMPUTES COSINE SUMS, CT(J) J=JMIN,,JMAX, ON	0015
			TWC INPUT SERIES, SS(I) AND AS(I) $I=0,1,\ldots,I$ , ACCORDING	0016
	ТO L			0017
		SUM ( SS(I) * COS(I * J*(PI/M)) )	J EVEN	0018
	$I = 0$ $CT(J) =$			0019 0020
	Ł			0021
		SUM ( AS(I)*COS(I*J*(PI/M)) )	J ODD	0022
	$I = 0$			0023
				0024
		FCR $J = JMIN, JMIN+1, \ldots, JMAX$		0025
	<b>WHERE</b>			0026
	$PI = 3.14159265$			0027
		$M = INPUT PARAMETER$		0028
			$COS(I*(PI/M))$ $I=0,1,,M$ IS AN INPUT TABLE SS(I), AS(I), MAY BE EITHER FIXED OR FLOATING POINT	0029 0030
			(THE COSINE TABLE MUST CORRESPOND IN TYPE)	0031
		C LSTHN= JMIN LSTHN JMAX LSTHN= M		0032
۰				0033
٠	SISP COMPUTES SINE SUMS, ST(J)			0034
۰	L			0035
		SUM ( AA(I)*SIN(I*J*(PI/M)) )	<b>J EVEN</b>	0036
۰ ۰	$I = 0$ $ST(J) =$			0037 0038
	L			0039
۰		SUM ( SA(I)*SIN(I*J*(PI/M)) )	J ODD	0040
	$I = 0$			0041
				0042
٠		FCR $J = JMIN, JMIN+1, \ldots, JMAX$		0043
	WHERE			0044
٠			$SIN(I+(PI/M))$ $I=0,1,,M$ IS AN INPUT TABLE AA, SA, AND THE SINE TABLE ARE FIXED OR FLOATING	CO45 0046
				0047
			COSISP COMPUTES BOTH CT(J) AND ST(J) AS DEFINED ABOVE	0048
				0049
			NOTE THAT THE FUNDAMENTAL FREQUENCY AS DEFINED BY THE	0050
۰		INPUT TABLES HAS PERIOD = EVEN NO. OF POINTS = 2M		0051
٠				0052
	* EQUIPMENT - 709 CR 7090 (MAIN FRAME ONLY)	* LANGUAGE - FAP SUBROUTINE (FORTRAN II COMPATIBLE)		0053
STORAGE - 492 REGISTERS				0054 0055
* SPEED		709-FIXED PT 709-FLOATING PT		0056
٠	<b>COSP</b>		MACHINE CYCLES	0057
۰	<b>SISP</b>	$34*K*(L+1)$ $37*K*(L+1)$ $39*K*(L+1)$ $43*K*(L+1)$	MACHINE CYCLES	0058
۰		$CCSISP 67*K*(L+1) 72*K*(L+1)$	MACHINE CYCLES	0059
$\bullet$	WHERE $K = JMAX-JMIN+1$			0060
٠		(REDUCE ESTIMATES ABOUT IC PERCENT FOR 7090)		0061
* AUTHOR	$-$ S.M. SIMPSON, OCT 26, 61			0062
۰ $\bullet$	$---USAGE---$			0063 0064
$\bullet$				0065
	* TRANSFER VECTOR CONTAINS ROUTINES - NONE			0066
۰	AND FORTRAN SYSTEM ROUTINES -	NONE		0067
				0068
٠				0069
* FORTRAN USAGE OF COSP				
۰		CALL COSP (SSX, ASX, L, COSTAB, M, JMIN, JMAX, TYPE, COSTR)		0070
٠ + INPUTS TO COSP				0071 0072

 $\Delta \sim 10^{11}$ 

```
221
```


```
222
```








## **22)**



### 22c



 $227$ 



 $\frac{1}{\sqrt{2}}$ 



294 **NOP** 0605 **195 NOP** 0606 **STO** SMCD 0607 \*INCREMENT INDEX FOR B 0608 TRAVEL, BY -JE FO 0609 2100 TXI  $+1,4,4$ 0610 **REVERSE)** \*CHECK IF INDEX HAS RUN CFF END (GREATER THAN M FOR 0611 FORWARD TRAVEL, LESS THAN ZERO FOR REVERSE)<br>(HOWEVER FOR REVERSE TRAVEL XR4 GOING NEGATIVE MEANS 0612 0613  $\bullet$ XR4 GETS GREATER THAN M, SO SAME TEST APPLIES) 0614 **Z101 TXH**  $2120, 4, **$  $...$ 0615 \*INCREMENT INDEX FOR ODD HARMONICS (BY+JO OR -(JO)) 0616 AND MAKE SAME KIND OF END TEST 0617 Z102 TXI<br>Z103 TXH (\*\*\*JO FORWARD)  $+1,2,++$  $(**=-JD REVERSE)$ 0618  $2110,2,***$  $[AB=M]$ 0619 \*INCREMENT DATA INDEX BY 1 AND CHECK FOR END OF DATA 0620 LOOPING BACK TO PLACE DETERMINED BY WHETHER COSP OR 0621 SISP OR COSISP AND FORWARD OR BACKWARD AND EVEN OR ODD 0622 Z104 TXI  $***1,1,1$ 0623 2105 TXL  $4 + A = 01$  $.........$  $(1XL + A, 1, 1B)$ 0624 \*\*A=290 FOR COSP 0625 FOR SISP OR COSISP (INITIAL = 250)  $\bullet$ 0626 \*\*A=250 EVEN AND ODD HARMONICS FORWARD 0627  $\bullet$ \*\*A=260 EVEN FORWARD, ODD REVERSE<br>EVEN REVERSÉ, ODD FORWARD 0628  $+44 = 270$ 0629 \*\*A=Z80 EVEN AND ODD REVERSE 0630 (\*\*=Z107 FOR COSP OR COSISP, 2106 TRA 0631 \*\*\*Z115 FOR SISP) 0632 \*READJUSTMENTS WHEN COD HARMONIC INDEX RUNS OFF END 0633 \*FORWARD OR BACKWARD 0634  $2110$  ZET **SWC** 0635 TRA  $2113$ **BACKWARD** 0636 **CLA** K1 0637 SWC ST<sub>0</sub> 0638 \*IF FCRWARD SET TO GC BACKWARD ON ODD 0639 TEMP,2 **Z111 SXD** 0640 **CLA** 2 M 0641 TEMP **SUB** 0642 PDX  $0, 2$ 0643 CLA MJC 0644 510 2102 0645 \*IF COSP GO BACK, IF NOT REMAKE FORK AT Z105 0646 COSP SISP OR COSISP 0647 **NOP** ZET SWEI 2112 (TRA 2104 OR 0648 TRA **Z112A** 0649 CLA (KA10 = PZE 260) **KA10** 0650 **2105 STA** 0651 TRA 2104 0652 **Z112A CLA KA12** {KA12=PZE Z80} 0653 **STA** 2105 0654 TRA 0655 2104 \*IF BACKWARDS SET TO GO FORWARDS ON ODD 0656 SWC Z113 STZ 0657 PXA  $0, 2$ 0658 PAC 0659  $0 - 2$ CLA JC. 0660 STD. **Z102** 0661 \*IF COSP GO BACK, IF NOT REMAKE FORK AT 2105 0662 SISP OR COSISP **COSP** 0663 2114 **NOP** (TRA Z104 OR ZET SWEI 0664 TRA Z114A 0665 CLA **KA9** (KA9=PZE Z50) 0666 **STA** 2105 0667 0668 **TRA** 2104 (KA11=PZE 270) 0669 **Z114A CLA KA11** 0670 **STA** 7105 TRA **Z1C4** 0671 \*READJUSTMENT WHEN EVEN HARMONIC INDEX RUNS OFF END 0672 \*WHICH WAY WERE WE GOING 0673

\*\*\*\*\*\*\*

........

 $\bullet$ 

(PAGE 9)

0599

0600

0601

0602

0603

0604

COSP

\*

LCQ

**NCP** 

**NOP** 

STC

LDQ

 $***,4$ 

SMCE

 $***,2$ 

THIS PART IS FOR

COSP

290

Z91

292

Z93

\*\*\*\*\*\*\*\*\*\* (PAGE 9)



#### 2 **f**



## $\alpha$



 $\ddot{\phantom{a}}$ 









```
235
```








 $\sim 10^6$ 







 $\ddot{\phantom{a}}$ 

# ~i4i



```
\bar{2}42
```


 $\hat{\mathcal{A}}$ 



 $\mathcal{L}^{\text{max}}_{\text{max}}$ 

 $\ddot{\phantom{0}}$ 

```
2PROGRAM LISTINGS1
```






# 2 **c'**



```
24\%
```






 $\mathcal{L}(\mathcal{A})$  .









```
252
```

```
253
```


```
254
```
PROGRAM LISTINGS \*\*\*\*\*\*\*\*\*\*\*\*\* **LI\*\*\*\*\*NTR\*\*\*\*\*\*\*\*\*\*\*\*\*\*** \* LINTR1<br>\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* **\*** LINTRI \* \*\*\*\*\*\*\*\*\*\*\*\* \* LINTR1 (SUHRCUTINE) 2/18/63 **LAST** CARD **IN** DECK IS **NO.** 0092 **\*** LABEL 0001 CLINTR1 0002 SUBROUTINE LINTR1(X, XLO, DELX, TABLE, NTABLE, YOFX) **0003** C 0004 C  $---ABSTRACT---$ **0005 0006 C** TITLE **-** LINTR1 **0007** C LINEAR INTERPOLATION IN A TABLE **0008** 0009 **0010**  $\mathbf{C}$ LINTR1 **INTERPOLATES LINEARLY IN A** TABLE TO **FIND** A **VALUE**  $\mathbf c$ **0011** WHICH LIES BETWEEN THE TABULATED VALUES. XLO **IS** THE ARGUMENT CCRRESPONDING TO THE LOWEST TABULATED VALUE. DELX  $\mathbf c$ 0312 IS THE ARGUMENT DIFFERENCE BETWEEN TABULAR VALUES.  $\mathbf c$ **0013** C THE TABLE IS LOCATED IN TABLE(I). X **IS** THE ARGUMENT AND 0014 C YOFX IS THE INTERPOLATED VALUE. HENCE 0015  $\mathsf{C}$ 0016  $\mathbf{C}$ **0017** XTRA  $\mathsf{C}$ YOFX = TABLE(L) **+** (TABLE(L+1) - TABLE(L)) **0018**  $\mathbf c$ DELX 0019  $\tilde{c}$ 0020 WHERE L IS SUCH THAT 0021  $\tilde{c}$ XLO+(L-1)\*DELX LSTHN= X LSTHN XLO+L\*DELX 0022 **AND** XTRA = X-XLO-(L-1)\*DELX **0023** 0024 C DELX IS **CONSTRAINED** TO **BE** POSITIVE **0025** C X **MUST** LIE IN THE ARGUMENT RANGE OF THE TABLE.  $\mathsf{C}$ **0026**  $\mathbf c$ 0027 **LANGUAGE** - FORTRAN II SUBROUTINE 0028 **0029** EQUIPMENT STORAGE **-** 709 OR 709C (MAIN FRAME ONLY) **- 96** REGISTERS **0030**  $\mathsf{c}$ **SPEED** 0031  $\mathbf{C}$ AUTHOR *-* **S.** P. SIMPSON **0032** 0033  $\mathbf c$ C ---- USAGE---- 0034  $\mathbf c$ **0035** TRANSFER VECTOR CCNTAINS ROUTINES - NONE **0036** C AND FORTRAN SYSTEM ROUTINES **-** NONE **0037** C. **0038**  $\epsilon$ FORTRAN USAGE C. **0039** CALL LINTRI(X,XLO,DELX,TABLE,NTABLE,YOFX) C 0040  $\mathbf c$ 0041 C INPUTS 0042  $\mathbf{C}$ 0043  $\mathsf{C}$ X **IS** ARGUMENT FOR WHICH INTERPOLATION IS DESIRED. 0044  $\mathsf{C}$ XLC LSTHN OR = X LSTHN OR **=** XLO+(NTABLE-1)\*DELX. 0045  $\mathsf{C}$ 0046  $\mathsf{C}$ XLC IS THE ARGUMENT CORRESPONDING TO THE FIRST TABULAR 0047  $\mathbf c$ ENTRY. 0048  $\mathsf{C}$ 0049  $\mathsf{C}$ DELX **IS** THE ARGUMENT **DIFFERENCE BETWEEN** TWO SUCCESSIVE **0050**  $\mathbf c$ TABULAR ENTRIES. **0051**  $\mathsf{C}$ MUST EXCEED **C.0,** BUT THIS CONSTRAINT IS **NOT** CHECKED. **0052**  $\mathbf c$ **0053**  $\tilde{c}$ TABLE(I) **I=1...NTABLE IS A GIVEN** ARRAY **IN** WHICH **TABLE(J)** 0054  $\tilde{\mathbf{c}}$ CONTAINS Y(XLO+DELX\*(J-1)). **0055** c<br>c **0056** NTABLE IS THE LENGTH OF THE TABLE. **0057 0058** C **OUTPUTS 0059** C **0060** WILL CONTAIN THE LINEARLY INTERPOLATED **VALUE** C YOFX **0061** C **0062 C EXAMPLES 0063** C 0064 1. INPUTS -  $X=7.5$ **X=7.5** XLO=5. DELX=2.5 TABLE(1...9)=1.,4.,9., **0065** 16.,25.,36.,49.,64.,81. **0066** C **0067 C CUTPUTS** - YOFX=4. **0068** C **0069 C** 2. **INPUTS -** SAME **AS** EXAMPLE 1. EXCEPT X=21.3 **0070** C **OUTPUTS** - YOFX=56.8 **0071** C **C 3. INPUTS - SAME AS** EXAMPLE 1. EXCEPT X=25. **0072 C OUTPUTS** - YOFX=81. **0073** 0074



**\* MAXSN (SUBROUTINE) 2/18/63 LAST** CARD **IN** DECK **IS NO. 0169 \*** FAP **0001 \*MAXSN** 0002 **COUNT 150 0003** LBL **MAXSN** 0004 ENTRY MAXSN(LX+X+XMAX1+I)<br>ENTRY MINSN(LX+X+XMIN1+I) **0005**<br>
0006 ENTRY **MINSN (LX\*X\*XMIN1\*I)**<br>ENTRY MAXAB (LX\*X\*XMAX2\*I) ENTRY MAXAB (LX,X,XMAX2,I) **0007** ENTRY **MINAB** (LX,X,XMIN29I) **0008** \* **0009 \*** ---- ABSTRACT---- **0010 \* 0011 \*** TITLE **-** MAXSN , WITH SECONDARY ENTRY POINTS MINSN, MAXAB, **AND** MINAB 0012 \* FIND SIGNED OR UNSIGNED EXTREMAL VALUES OF A VECTOR. 0013  $*$  0014 \* MAXSN FINDS THE MAXIMUM SIGNED NUMBER, AND **ITS** INDEX, IN 0015 \* A VECTOR OF NUMBERS (EITHER FIXED OR FLOATING POINT). 0016 \* 0017 \* **MINSN** FINDS THE MINIMUM **SIGNED** NUMBER. 0018  $*$  0019 MAXAB FINDS THE MAXIMUM OF THE ABSOLUTE VALUES. 0020  $*$  0021 \* MINAB FINDS THE MINIMUM OF THE ABSOLUTE VALUES. 0022  $*$  0023 \* LANGUAGE - FAP SUBROUTINE (FORTRAN II COMPATIBLE) 0024 \* EQUIPMENT - **709** OR 7090 (MAIN FRAME ONLY) 0025 \* STORAGE - 54 REGISTERS 0026 \* SPEED - APPROX. 14N MACHINE CYCLES, N = LENGTH OF VECTOR = 0027<br>\* AUTHOR - J.F. CLAERBOUT - J.F. CLAERBOUT  $*$  0029  $---USAGE---$  0030  $*$  0031 \* TRANSFER VECTOR CONTAINS ROUTINES - NONE 0032 \* **AND** FORTRAN SYSTEM ROUTINES - NONE 0033  $*$  0034 \* FORTRAN USAGE FOR MAXSN 0035 \* CALL MAXSN (LX<sub>XX</sub>XMAX1,I) 0036<br>\* 0037  $\star$  0037 \* INPUTS 0038 \* **0039**  $X(1)$  I=1...LX IS A VECTOR OF NUMBERS. 0040 \* MAY BE FIXED OR FLOATING POINT.<br>\* 0042  $*$  0042 \* LX IS FORTRAN II INTEGER. 0043 MUST BE GRTHN=1. 0044<br>0045  $*$  0045 \* OUTPUTS 0046 **t\*** 0047 XMAX1 IS THE MAXIMUM SIGNED VALUE IN THE X VECTOR. THE STATE ROOMS  $*$  0049 \* I IS THE INDEX OF THE MAXIMUM SIGNED VALUE. Q050  $I_eE_e$   $X(I) = XMAX1$  0051 0052  $*$  0052 \* FORTRAN USAGE FOR MINSN<br>\* CALL MINSN (LX+X+XMIN1+I) \* CALL MINSN (LX,X,XMIN1,I) 0054<br>0055 0055  $\star$  0055 \* INPUTS SAME AS FOR MAXSN 0056  $\star$  0057 \* OUTPUTS 0058  $*$  0059 **S** XMIN1 IS THE MINIMUM SIGNED VALUE IN THE X VECTOR 0060

 $0061$  $\bullet$ IS THE INDEX OF THE MINIMUM SIGNED VALUE.  $\mathbf{I}$ 0062  $\bullet$ 0063 \* FORTRAN USAGE FOR MAXAB 0064  $\star$ CALL MAXAB (LX,X,XMAX2,I) 0065 0066  $\frac{M}{\sqrt{2}}$ **INPUTS** SAME AS FOR MAXSN  $\frac{1}{N}$ 0067 0068 **OUTPUTS**  $\pmb{\ast}$ 0069 0070  $\bullet$ XMAX2 IS THE MAXIMUM ABSOLUTE VALUE IN THE X VECTOR.  $0071$  $\frac{1}{N}$ NOTE THAT XMAX2 MAY BE NEGATIVE. 0072 0073  $\star$ IS THE INDEX OF THE MAXIMUM ABSOLUTE VALUE.  $\ddot{\bullet}$ 0074  $\mathbf{r}$ 0075  $\frac{1}{2}$ FORTRAN USAGE FOR MINAB 0076  $\ast$ CALL MINAB (LX,X,XMIN2,I) 0077 ×  $\star$ 0078 × **INPUTS** SAME AS FOR MAXSN 0079 0080  $\star$ **OUTPUTS** 0081  $\bullet$ 0082 IS THE MINIMUM ABSOLUTE VALUE IN THE X VECTOR. 0083 XMTN2  $\bullet$ NOTE THAT XMIN2 MAY BE NEGATIVE. 0084  $\ddot{\phantom{a}}$ 0085 IS THE INDEX OF THE MINIMUM ABSOLUTE VALUE. 0086  $\ast$  $\mathbf{I}$ 0087 × 0088  $\star$ **FXAMPLES** 0089  $\overline{\mathbf{r}}$ 1. INPUTS -  $X(1_{e+1}0) = -11_{e+}-8_{e+}-5_{e+}-2_{e+}1_{e+}4_{e+}$  7.10.13.13.16. 0090  $\bullet$ 0091  $\bullet$  $LX = 10$ CALL MAXSN (LX,X,XMAX1,I1) 0092  $\boldsymbol{\ast}$ USAGE  $\bullet$ CALL MINSN (LX,X,XMIN1,I2) 0093  $\bullet$ CALL MAXAB (LX,X,XMAX2,I3) 0094 ¥ CALL MINAB (LX,X,XMIN2,14) 0095 OUTPUTS - $XMAX1 = 16.$  $I1 = 10$ 0096  $\star$ 0097  $X M[N1 = -11$ .  $12 = 1$ 0098  $\bullet$  $XMAX2 = 16.$  $13 = 10$  $XMLN2 = 1.$  $14 =$ 0099 × 5 0100 圣  $2.5$  INPUTS -  $X(1.0010) = -16.0013.010.017.014.014.012.05008.011.$ 0101 ¥  $LX = 10$ <br>- SAME AS EXAMPLE 1. 0102  $\frac{1}{2}$ USAGE  $0103$ ÷ × OUTPUTS - $XMAX1 = 11.$  $11 = 10$ 0104  $\frac{12}{12} = \frac{12}{1}$  $XML = -16.$ 0105 ¥ 0106  $\boldsymbol{x}$  $XMAX2 = -16.$  $XMLN2 = -1.$  $14 = 6$ 0107  $\bullet$ 0108  $\ddot{\bullet}$ 3. INPUTS -  $X(1...10) = -16,-13,-10,-7,-4,-1,2,5,8,11$  LX = 10 0109  $\bullet$ - SAME AS EXAMPLE 1. **USAGE** 0110  $\bullet$ 0111 **OUTPUTS**  $XMAX1 = 11$  $11 = 10$  $\bullet$  $XML = -16$ 0112  $12 = 1$  $\ast$  $\ddot{\phantom{1}}$  $XMAX2 = -16$  $13 = 1$ 0113  $XMIN2 = -1$  $14 =$ 0114  $\star$ 6 0115  $\bullet$ **HTR**  $\mathsf{o}\,$ 0116 1.MAXSN 0117 **BCT** 0118 MAXSN CLA **MX STO** USE 0119 0120 **TRA**  $*+3$ 0121 MINSN CLA **MN** 



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250
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# $26i$ PROGRAM LISTINGS



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262
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 $\label{eq:1} \frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1}{\sqrt{2}}\sum_{i=1}^n\frac{1$ 



PROGRAM LISTINGS \* \*  $\bullet$ NOINT1  $\bullet$ \* \* IS FLOATING POINT # FORTRAN USAGE OF NCINT2 CALL NCINT2(XMEAN, XSD, NDIV, XDIV, IANS) \* INPUTS TO NOINT2 = MEAN OF X SERIES = STANDARD DEVIATION OF X SERIES. MUST BE GRTHN 0. = NUMBER OF EQUALLY LIKELY DIVISIONS INTO WHICH XSERIES THE SECOND IS FROM XDIV(1) TO XDIV(2) ETC. THE LAST

(PAGE 2)

0075

0076 0077

0078

0079

 $0080$ 

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0086

#### 0087 NDIV IS TO BE PLACED.<br>MUST BE GRTHN 1 0088 0089 0090 0091 QUTPUTS FROM NOINT2 0092 I=1...NDIV-1 ARE THE X VALUES FOR EQUALLY LIKELY 0093 XDIV(I) DIVISIONS. FIRST DIVISION IS FROM -INFINITY TO XDIV(1), 0094 0095 DIVISION IS FROM XOIV(NOIV-1) TO +INFINITY. 0096 0097 0098  $=0$  NCRMAL IANS 0099 ILLEGAL XSD  $\approx 1$ 0100  $= 2$ **ILLEGAL NDIV** 0101 0102 \* EXAMPLES OF NOINT1 0103 1. INPUTS -  $X=-5$ .<br>CUTPUTS - PROB=0. 0104 0105 0106 0107 2. INPUTS -  $X=-4$ . CUTPUTS - PRCB=.32 E-04 0108 0109 3. INPUTS -  $X = .013$ <br>OUTPUTS - PRCB=.5052 0110 0111 0112 4. INPUTS -  $X=4$ .<br>CUTPUTS - PRCB=.999968 0113 0114 0115 5. INPUTS -  $X=4.1$ <br>CUTPUTS - PROB=1. 0116 0117 0118



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NOINT1

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**XMFAN** 

XSD



#### $26\%$



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 $\bar{\beta}$ 



PROGRAM **LISTINGS**

\* **NOINT2** REFER TO NOINT1 \* **NOINT2 \*** REFER TO **NOINTI**

 $\hat{\mathcal{A}}$ 







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272
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275
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277
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\*.,.+\*\*\*,\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* PROGRAM LISTINGS \* \* PROB2 PROB2 (PAGE 2) (PAGE 2) C **=-6** ILLEGAL IXHI. IXHI GRTHN **26** OR LSTHN 1. 0075 C =3 JCB DONE BUT N=0 AND ONLY CONTRIBUTIONS TO P(I,J) ARE 0076<br>C = CN THE DIAGONAL. C CON THE DIAGONAL. CONSIDERING COMPANY ON THE DIAGONAL. **C** 0078 **C EXAMPLES 0079 C** 0080 **C 1. INPUTS -** IX(I)=0, LX=5, **N=v1** IXHI=5 **0081** C CUTPUTS **-** IP(IJ)=O , P(I,J)=O **,** IANS=-1 0082 **C** 0083 **C** 2. **INPUTS -** SAME AS EXAMPLE 1 EXCEPT IX(I)=l,2,3,4y6 0084 **C** CUTPUTS - SAME AS EXAMPLE 1 0085<br>
C CUTPUTS - SAME AS EXAMPLE 1 0086 **C 0086 C 3, INPUTS -** SAME AS EXAMPLE 2 EXCEPT LX=O 0087 C CUTPUTS - IANS=-2 0088<br>C CUTPUTS - IANS=-2 0088 C **0089** C 4. INPUTS - SAME AS EXAMPLE 2 EXCEPT IXHI=O 0090 **C CUTPUTS** - IAKS=-6 **0091** C 0092 **C 5. INPUTS -** SAME AS EXAMPLE 4 EXCEPT IXHI=26 0093 **C** CUTPUTS - IANS=-6 0094<br>
C C  $\mathsf C$  , and the contract of **C 6. INPUTS -** SAME AS EXAMPLE 2 EXCEPT IX(5)=5, N=-6 **0096**  $\frac{1}{2}$  CUTPUTS - IANS=-3 0097 **C** 0.098 **C 7. INPUTS -** IX(I)=1,1,2,2,3,3,4,4,5,5,1,2,2,3,45,5,1,1,1,1,1,1,1 **0099** C  $I X H I = 5, L X = 21, N=1$  0100<br>C CUTPUTS - IANS=0 0101 **C CUTPUTS** - IANS=O 0101 **C** 4 2 0 **0 0** .2 **.1** .0 .0 **.0** 0102 **C** 0 2 2 0 0 .C .1 .1 **.0** .0 **0103** C IP(I,J)= 0 0 1 2 0 P(I,J)= .0 .0 .05 **.1** .0 0104 C 0 0 0 1 2 .0 **.0** .0 **.05** .1 **0105** C 2 **0** 0 0 2 .1 **.0** .0 .0 .1 0106 C 0107<sup>1</sup> **C 8. INPUTS -** SAME **AS** EXAMPLE **7** EXCEPT N=-1 0108 C OUTPUTS - IANS=O 0109 **C** 0110 **C** 4 0 0 0 2 .2 .0 .0 .0 .1 0111 C 2 2 0 **0** 0 .1 **.1** .0 **.0** .0 0112 C  $IP'(1, J) = \begin{bmatrix} 0 & 2 & 1 & 0 & 0 \\ 0 & 0 & 2 & 1 & 0 \\ 0 & 0 & 0 & 2 & 1 \end{bmatrix}$   $P(1, J) = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \end{bmatrix}$   $P(1, J) = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 \end{bmatrix}$ **C** 0 0 2 1 0 .0 **.0** .1 .05 .0 0114 **C** C 0 0 2 2 .0 **.0 .0** .1 .1 **0115** C 0116 **C 9.** INPUTS **-** SAME AS EXAMPLE **7** EXCEPT LX=24, N=3 **0117 C CUTPUTS** - IANS=0O 0118 **C 3 1** 2 **0** 0 **.15 .05 .1 .0** .0 0119 **C** 0 0 1 2 1 .0 .0 **.05 .1 .05** 0120 C IP(I,J)= 0 0 0 1 2 P(I,J)= .0 .0 .0 .05 .1 0121 C 2 0 0 0 1 .1 .0 .0 .0 **.05** 0122 C 2 2 0 0 0 .1 .1 .0 .0 .0 0123 C 0124 C10. INPUTS **-** SAME **AS** EXAMPLE **7** EXCEPT LX=20, **N=O** 0125 **C OUTPUTS** - IANS=3 **0126**  $\sim$  0127 C **6** 0 0 0 0 .3 .0 .0 **.0** .0 0128 **C 0** 4 **0 0 0 .0** .2 **.0 .0 .0 0129**  $\overline{C}$  **IP(I,J)=** 0 0 3 0 0 01310<br>  $\overline{C}$  **IP(I,J)=** 0 0 0 3 0 0 0 15 0 0131<br>  $\overline{C}$  0 0 0 0 3 0 0 0 0 0 0 0131 **C 0 0** 0 3 0 **.0 .0** .0 .15 .0 0131 C 0 0 **0 0** 4 **.0 .0 .0 .0** .2 **0132** C **0133** DIMENSION IX(1000),IP(25,25),P(25,25) 0134 C **CHECK** LX **0135**  $IANS=-2$  0136 IF(LX) **9999,9999,2 0137** 2 IANS=-6 **0138** C **CHECK** IXHI **0139** IF(IXHI) **9999,9999,3** 0140 **3** IF(IXHI-25) 4,4,9999 0141 CHECK IX SERIES **CHECK IX SERIES** 20142<br>4 IANS=-1 0143 4 IANS=-l 0143  $\begin{array}{ccc}\n 0 & 1 & 1 & -1,1X \\
 0 & 1 & 1 & 0144 \\
 \hline\n 1 & 0 & 145\n \end{array}$ IF(IX(I)) 9999,9999,11 0145<br>IF(IX(I)-IXHI) 1.1.9999 0146 11 IF(IX(I)-IXHI) 1,1,9999 0146<br>1 CONTINUE 0147 1 CONTINUE 20147  $IANS=-3$  0148

**C CHECK N** 0149



 $\label{eq:2.1} \frac{1}{\sqrt{2}}\left(\frac{1}{\sqrt{2}}\right)^{2} \left(\frac{1}{\sqrt{2}}\right)^{2} \left(\$ 

#### BIBLIOGRAPHY

Blackman, R. B., and Tukey, J. W., 1958, The measurement of power spectra: Dover Publications Inc., New York

Bogert, B. P., 1961, The transfer function of a short-period vertical seismograph: Bull. Seis. Soc. Am., 51:503-513.

Bullen, K. E., 1953, An Introduction to the Theory of Seismology: Cambridge University Press, Cambridge.

Claerbout, J. F., 1963, Digital Filters and Applications to Seismic Detection and Discrimination: S.M. Thesis, Mass. Inst. of Tech.

Cramer, H., 1946, Mathematical Methods of Statistics: Princeton Univ. Press.

Davenport, W. B., Jr., and Root, W. L., 1958, An Introduction to the Theory of Random Signals and Noise: McGraw-Hill Book Company, Inc., New York.

DeVorkin, D., **1963,** Representation Schemes for Investigating Non-Linear Processes: Ph.D. Thesis, Mass. Inst. of Tech.

Doob, J. L., 1953, Stochastic Processes: John Wiley and Sons, New York.

Geotechnical Corporation, 1961, Personal communication.

Haq, K. E., 1954, The nature and origin of microseisms: Sc.D. Thesis, Mass. Inst. of Tech.

Kenyon, K., 1961, Microseisms and Water Waves: S.M. Thesis, Mass. Inst. of Tech.

Kolmogorov, A., **1939,** Sur l'interpolation et extrapolation des suites stationnaires: C. R. Acad. Sci., V. 208, Paris, 2043-2045.

Lamb, H., 1932, Hydrodynamics: Dover Publications, New York.

Lanczos, C., 1956, Applied Analysis: Prentice Hall, Inc., New Jersey.

Lee, Y. W., 1960, Statistical Theory of Communication: John Wiley and Sons, Inc., New York.

Levinson, N., 1949, The Wiener RMS (Root Mean Square) Error Criterion in Filter Design and Prediction: Appendix B of Wiener (1949).

Longuet-Higgins, M. **S., 1950,** A theory of the origin of microseisms: Phil. Trans. Roy. Soc. London **(A).**

Longuet-Higgins, M. **S.,** and Ursell, F., Sea waves and microseisms: Nature.

Ramirez, **J. E.,** 1940, An experimental investigation of the nature and origin of microseisms at St. Louis, Missouri: Bull. Seis. Soc. Am.

Rand Corporation, **1955, A** million random digits with **100,000** normal deviates: Glencoe, **Ill.,** Free Press.

Robinson, **E. A.,** 1954, Predictive Decomposition of Time Series with Applications to Seismic Exploration: Ph.D. Thesis, Mass. Inst. of Tech.

Robinson, **E. A., 1962,** Random Wavelets and Cybernetic Systems: No. Nine of Griffin's Statistical Momographs and Courses, Charles Griffin and Company Limited, London.

Romney, **C., 1959,** Amplitudes of seismic body waves from nuclear explosions. **J.** Geophys. Res. V. 64, 1489-1498.

Simpson, **S.** M., Robinson, **E. A.,** Claerbout, **J.** F., Galbraith, **J. N.,** Clark, **J.,** 1961a, Initial studies on underground nuclear detection with seismic data prepared **by** a novel digitization system; Annual Report No. 1 of Contract **AF 19(604)7378** at M.I.T., prepared for Geophysics Research Directorate, Air Force Cambridge Research Laboratory, Office of Aerospace Research, United States Air Force Bedford, Mass.

Simpson, **S.** M., Robinson, **E. A.,** Claerbout, **J.** F., Galbraith, **J. N.,** Ross, W. P., Clark, **J., 1961b:** Time series techniques applied to underground nuclear detection and further digitized seismic data: Scientific Report No. 2 of Contract **AF 19(604)7378** continued as above.

Simpson, **S.** M., Robinson, **E. A.** Claerbout, **J.** F., Clark, **J.,** Galbraith, **J.** N., Pan, **C.,** Wiggins, R., 1962a, Continued numerical studies on underground nuclear detection and further digitized seismic data; Scientific Report No. **3** of Contract **AF 19(604)7378** continued as above.

Simpson, **S.** M., Robinson, **E. A.,** Claerbout, **J.** F., Clark, **J.,** Galbraith, **J. N.,** Greenfield, R. **J.,** Wiggins, R. **A., 1962b,** Magnetic tape copies of M.I.T. Geophysics Program Set I, (Time Series Programs for the IBM **709, 7090);** Scientific Report No. 4 of Contract **AF 19(604)7378** continued as above.

Simpson, **S.** M., Jr., (Director M.I.T. Geophysical Analysis Group), **1955,** Linear Operators and Seismic Noise: MIT **GAG** Report No. **9**

Simpson, **S.** M., Jr., **1956,** Properties, Origin and Treatment of Certain Types of Seismic Noise: MIT **GAG** Report **No.** 10a.

Stoker, **J. J., 1957,** Water Waves: Interscience Publishers, Inc., New York.

Vesiac Staff, **1962,** Problems in Seismic Background Noise: **VESIAC** Advisory Report 4410-32-X, Acoustics and Seismics Laboratory, Institute of Science and Technology, The University of Michigan.

Wadsworth, **G.** P., and Bryan, **J. G., 1960,** Introduction to Probability and Random Variables: McGraw-Hill Book Company, Inc., New York

Wadsworth, **G.** P., Robinson, **E. A.,** Bryan, **J. G.,** and Hurley, P. M., **1953,** Detection of Reflections on Seismic Records **by** Linear Operators: Geophysics, V. **18, No. 3,** July **1953.**

Whittle, P., **1963,** Prediction and Regulation; (in press).

Wiener, **N.,** 1949, Extrapolation, Interpolation, and Smoothing of Stationary Time Series: The Technology Press of the Mass. Inst. of Tech., and John Wiley and Sons, Inc., New York.

Wold, H., **1938, A** study in the analysis of stationary time series: Uppsala. Almqvist and Wiksells.

BIOGRAPHICAL NOTE

The author was born in Philadelphia, Pennsylvania on April 26, 1936. He attended the Germantown Friends School in Philadelphia from 1941 until his graduation in 1954. He entered the Massachusetts Institute of Technology in 1954 and obtained a Bachelor of Science degree in Physics in 1958. He entered the Graduate School at M.I.T. in the Geology and Geophysics Department in 1958 and was a research assistant under Professor W. F. Brace until 1960. He then held a tuition scholarship for a year and research assistantships for two years while working for Professor S. M. Simpson, Jr on this thesis. He was married in 1960 to the former Miss Joan Blumenstiel of Alliance, Ohio.