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SEISMIC ARRAY PROCESSING TECHNIQUES

Quarterly Report No. 3

15 February 1970 through 15 May 1970

Frank H. Binder, Program Manager
Area Code 214, 238-6521

TEXAS INSTRUMENTS INCORPORATED
Services Group
P.O. Box 5621
Dallas, Texas 75222

Contract No. F 33657-70-C-0100
Amount of Contract: $339,052
Beginning 15 July 1969
Ending 14 July 1970

Prepared for

AIR FORCE TECHNICAL APPLICATIONS CENTER
Washington, D. C. 20333

Sponsored by

ADVANCED RESEARCH PROJECTS AGENCY
Nuclear Test Detection Office
ARPA Order No. 624
AFTAC Project No. VELA/T/0701/B/ASD

3 June 1970

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TEXAS INSTRUMENTS
INCORPORATED
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SERVICES GROUP

3 June 1970

Headquarters
United States Air Force
AFTAC/VSC
Washington, D. C. 20333

Attention: Lt. John Woods

Subject: Third Quarterly Report Covering Period
15 February 1970 through 15 May 1970

Identification: AFTAC Project No.: VELA/T/0701/B/ASD
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SECTION I: INTRODUCTION

Work done this quarter is divided into five general categories:

- Techniques for designing fixed filters (array velocity filters) off-line
- On-line adaptive processing techniques
- Long-Period array data analysis
- TFO short-period array data analysis
- Hi Resolution wavenumber spectra displays of seismic array data

Sections II through VI summarize work progress under each of the categories.

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SECTION II: DESIGN OF FIXED FILTERS

Site 3

A. TIME-DOMAIN ADAPTIVE MCF DESIGN

Five Site 3 noise samples were chosen for adaptive MCF design. Each sample was divided into 256 point segments and the first segment was adaptively processed. Then a second segment was selected from each sample and was processed, proceeding through the samples with the sequence reversed from that of the first segments. Two more segments from each sample were then processed using this procedure, for a total of four segments processed from each of the four noise samples. The adaptive processing was begun with the updating of the filters set at 37.5 per cent of the estimated divergence rate; the percent rate was decreased as an exponential function of the number of data points processed until a rate of one percent was reached at the end of the data. The resultant multichannel filter was evaluated by applying it to ensemble and non-ensemble data.

To determine the additional noise rejection achievable by fine tuning, the above design procedure was repeated, beginning with the above filters and a one percent rate and ending with a 0.01 percent rate. The fine tuning resulted in negligible noise rejection improvement below 1.0 Hz and from 0.5 to 1.0 db improvement above 1.0 Hz.

The noise rejection capability of the adaptively designed MCF was compared to the optimum maximum-likelihood MCF which had been designed from Site 3 data in the convergence rate study. The two filters were applied to ensemble noise data and power density spectra were computed for the input noise (using a reference input channel) and the output noise from the MCF. The power in each of three frequency bands, 0-1.6 Hz, 1.6-3.4 Hz, and 3.4-5.1 Hz was determined from each spectrum by integrating the power density over these frequency intervals. The difference in db between the input and output noise power in each frequency range gives a measure of the noise rejection capability of the filter in each frequency range. The noise rejection of the maximum-likelihood MCF in the three frequency ranges was better than that of the adaptively designed MCF by 0.3, 2.1, and 2.2 db, respectively.

B. CONVENTIONAL FREQUENCY DOMAIN WIENER MCF DESIGN

For purposes of comparison with adaptive MCF design and application, Wiener MCFs were designed in the frequency domain and applied in the time domain. Effects of gain differences among channels in the frequency domain MCF design were eliminated by constraining the signal to have the same relative gains channel-to-channel as the noise.

Weighting Study

The use of weighting functions in the design of frequency domain MCFs was investigated using four weighting methods and then comparing the results to unweighted frequency domain designed and to time domain designed MCFs. The first weighting scheme consisted of weighting the response of each channel by the average power spectra of the design matrix. The second weighting scheme was similar to the first except that a "triangular" method[(1)] was used in obtaining the multichannel time domain operators.

Time domain evaluation of these two methods showed that some of the spectral lines in the data were not being adequately rejected. Therefore, a variation of the first method was employed, in which weights for the frequencies at which the spectral lines occurred were increased before the time domain operators were obtained. Some improvement was obtained but the results were still 2 to 5 db inferior to the time-domain maximum-likelihood MCF.

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(1) Texas Instruments Incorporated, 1970: Suboptimal Multichannel Digital Filters, Seismic Array Processing Techniques Tech. Rpt. 2, Contract F33657-70-C-0100, 28 Jan.

The last method attempted to concentrate on noise rejection in the 1.6-3.4 Hz frequency band. The weights in this band, obtained by the last method above, were increased first by a factor of 10 and then by 100. For each set the time domain operators were obtained by each of the first two weighting methods described above. For both factors the first weighting method (single channel at a time) yielded about 1 db improvement over the best of the other methods in the 1.6-3.4 Hz band; the results for the triangular method were nearly identical with the others.

All the above methods used MCF responses obtained at 65 frequencies; the final time-domain operators used were all truncated to 29 points.

Truncation Study

In order to evaluate the effect of truncating the time domain operators output from standard frequency domain MCF designs the filters were evaluated on the data in their full length and in two truncated forms. First the total length 129 point operators (from 65 frequencies) were used; this filter set was then truncated to 65 and 29 points and evaluated at each length. The 65 point set performed within 1 db of the 129 point set; the 29 point set resulted in 2-3 db loss in noise rejection capability.

Transform Gate Length Study

Included in the investigation of frequency domain MCF design was a study of the effect of the gate length used in the transformation of the time series data to the frequency domain. Five Site 3 noise samples were transformed using 512 point, 256 point, 64 point, 32 point, and 16 point segments of data in computing the crosspower matrices which were stacked to form the design matrix. When the resulting time domain operators were evaluated, the results showed approximately 1.5 db loss in noise rejection capability between the 512 point and 64 point transforms, and 4-5 db between the 512 point and 16 point transforms.

Gain Difference Effects

Part of the observed superior noise rejection capability of the time domain designed maximum-likelihood MCF could have resulted from utilization of gain inequalities among the array channels. To investigate this possibility, MCFs were designed in the frequency domain with the signal power set equal to the average noise power over all channels, a condition corresponding in the frequency-domain filter design to the condition of utilization of gain inequalities by the time-domain designed filter.

These frequency-domain-designed filters were applied to Site 3 data and the results compared to those obtained with the filters designed with the constraint of equal signal and noise gains for each channel. The results show that the noise rejection improves about 1.5 db for the condition corresponding to utilization of gain inequalities in the time domain design. This is still about 4.5 db poorer than the performance of the filter system designed in the time-domain.

Since the time-domain maximum-likelihood MCF had performed very well in rejecting noise, this filter system was applied to two signals recorded at Site 3. The results showed about 3 db of signal attentuation of the signals at the MCF output compared to the signal on a reference channel. Less than 1 db signal attenuation would be expected on the basis of the array response of the MCF. Signal distortion was negligible.

The best frequency-domain and adaptive time-domain filters designed from Site 3 data will be applied to these signals and the results will be compared to the above results.

Site 2

A. TIME DOMAIN DESIGN AND APPLICATION

A study to process the Site 2 data in a manner paralleling the processing of the Site 3 data is being undertaken. Correlation matrices have been computed from five representative Site 2 noise samples, these will be stacked and input to a program which will design a time-domain maximum-likelihood MCF from the stacked matrix. This filter will then be applied to the data.

B. FREQUENCY-DOMAIN DESIGN AND APPLICATION

Filters are presently being designed in the frequency-domain from the five Site 2 noise samples. Then will be applied in the time-domain and compared to the time-domain-designed filter results.

Details of the results described above and of the work in progress will be presented in a special report.

C. FREQUENCY-DOMAIN ADAPTIVE MCF DESIGN

The frequency-domain adaptive algorithm derived as shown in the last quarterly report has been used to process a set of synthetic stationary data[(2)]. Three filters with dimensions of 13-channel 29 points, 45 points, and 59 points have been designed. The filters do not converge very rapidly. The mean-square output decrease for the 29 point filter was 2.2 db after 1000 iterations of adaption; for the 45 point filter, 3 db after 3080 iterations; and for the 59 point filter, 3.2 db after 2800 iterations. For the best time-domain adaptive filter, the mean-square output decreased by 4.7 db after 3300 iterations. Thus, the frequency-domain adaptive filtering does not converge so fast as time-domain adaptive filtering.

The performance of the frequency-domain adaptively designed filters was evaluated by transforming them back to the time-domain and fixedly applying them to the time-domain data.

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(2) Special Report No. 3, Convergence of Time-Domain Adaptive Maximum-Likelihood Filters for Stationary Data, Texas Instruments Incorporated, 26 Feb.

The choice of convergence parameters in this experiment is based on trial-and-error and those parameters used may not be the best ones. This may effect the convergent rate of the adaptive algorithm. However, due to these discouraging results, the algorithm will not be applied to measured data. The detailed results and discussions of the experiment will be contained in a special report which is now being written.

SECTION III: ON-LINE ADAPTIVE MULTICHANNEL PROCESSING SIMULATION

The objective of this task is to simulate real-time adaptive on-line filtering and associated problems. This consists of the investigation of procedures for handling the following problems which could be encountered during continuous on-line processing.

- Degradation of filter performance resulting from signals
- Dead or noisy channels
- Degradation of performance of directional adaptive beams resulting from the digital sampling interval

This section describes the IBM 360 program specifications and plans for the completion of this task. Some preliminary results are also given.

A. SPECIFICATIONS OF ADAPTIVE MAXIMUM-LIKELIHOOD ON-LINE SIMULATION PROGRAM

An adaptive on-line simulation program has been written for the IBM 360. The filter output is given by

$$Y^t = \sum_{i=1}^{NC} \sum_{j=1}^{LF} F^t_{i,j} \; X_{i,t-j} \qquad (1)$$

where Y^t is the filter output at time t

$F^t_{i,j}$ is the filter weight of the i^{th} channel and the j^{th} point at time t

$X_{j,t}$ is the input data of the i^{th} channel at time t

NC is the number of channels

LF is the length of the filter

Filter coefficients are updated at each point in time according to the following maximum-likelihood algorithm.

$$F_{i,j}^{t+1} = F_{i,j}^{t} + \lambda^t Y^t (\overline{X}_{t-j} - X_{i,t-j}) \qquad (2)$$

where $\overline{X}_{t-j} = \frac{1}{NC} \sum_{i=1}^{NC} X_{i,t-j}$

λ^t is the rate of convergence parameter of time t

Filter coefficients must satisfy the constraints

$$\sum_{i=1}^{NC} F_{i,j} = \delta_{j,k} \quad j = 1, LF \qquad (3)$$

where $k = \frac{LF+1}{2}$ (LF must be odd).

If $F_{i,j}^1$ satisfies the constraints of equation (3) then the algorithm in equation (2) assures that $F_{i,j}^t$ will also satisfy the constraints for all t.

The simulation program also provides for scaling of λ^t in accordance with the magnitude of the input data, and also an exponential decay in the magnitude of λ^t.

$$\lambda^t = \frac{ZK_s (1-e^{-m/t})}{LF \sum_{i=1}^{NC} P_{i,t} + C} \qquad (4)$$

where $P_{i,t} = (1-\mu_1)(\overline{X}_t - X_{i,t})^Z + \mu_1 P_{i,t-1}$

$P_{i,1} = 0$

$0 \leq \mu_1 \leq 1$

B. PLANS FOR ADAPTIVE ON-LINE SIMULATION PROCESSING

Signal Effects

A question of importance in regard to an on-line processor is the effect of signals on the noise attenuation capabilities of the processor. Processing will be undertaken to measure the effects of signals of various S/N ratios and techniques to minimize these effects will be investigated.

A signal detection capability has been added to the simulation program whereby the filter coefficients will be frozen upon detection of a signal. Freezing when signal is present will prevent the filter from adapting to the signal and, therefore, the noise rejection of the filter will be preserved. Such a technique is expected to work well for large signals; however, small signals are not easily detected and will therefore probably degrade the filter performance somewhat. The magnitude of the filter degradation resulting from signals will be measured.

The signal detection scheme added to the simulation program freezes the filters if

$$\left| Y^t \right| > (N+Z)\, Q_{t-1} \tag{5}$$

In the processing completed to date, Q_t has been defined in two different ways

$$Q_t = (1-\mu_2)\, \overline{X}_t^{\,2} + \mu_2\, Q_{t-1} \tag{6}$$

or

$$Q_t = (1-\mu_2)\, Y_t^2 + \mu_2 Q_{t-1} \tag{7}$$

A noise sample has been processed with Q_i's computed according to equations (6) and (7) for various values of N. The purpose of this processing was to determine the minimum acceptable value of N. Not all values of N are acceptable since as N becomes small the filters are frozen on the noise even when signal is not present.

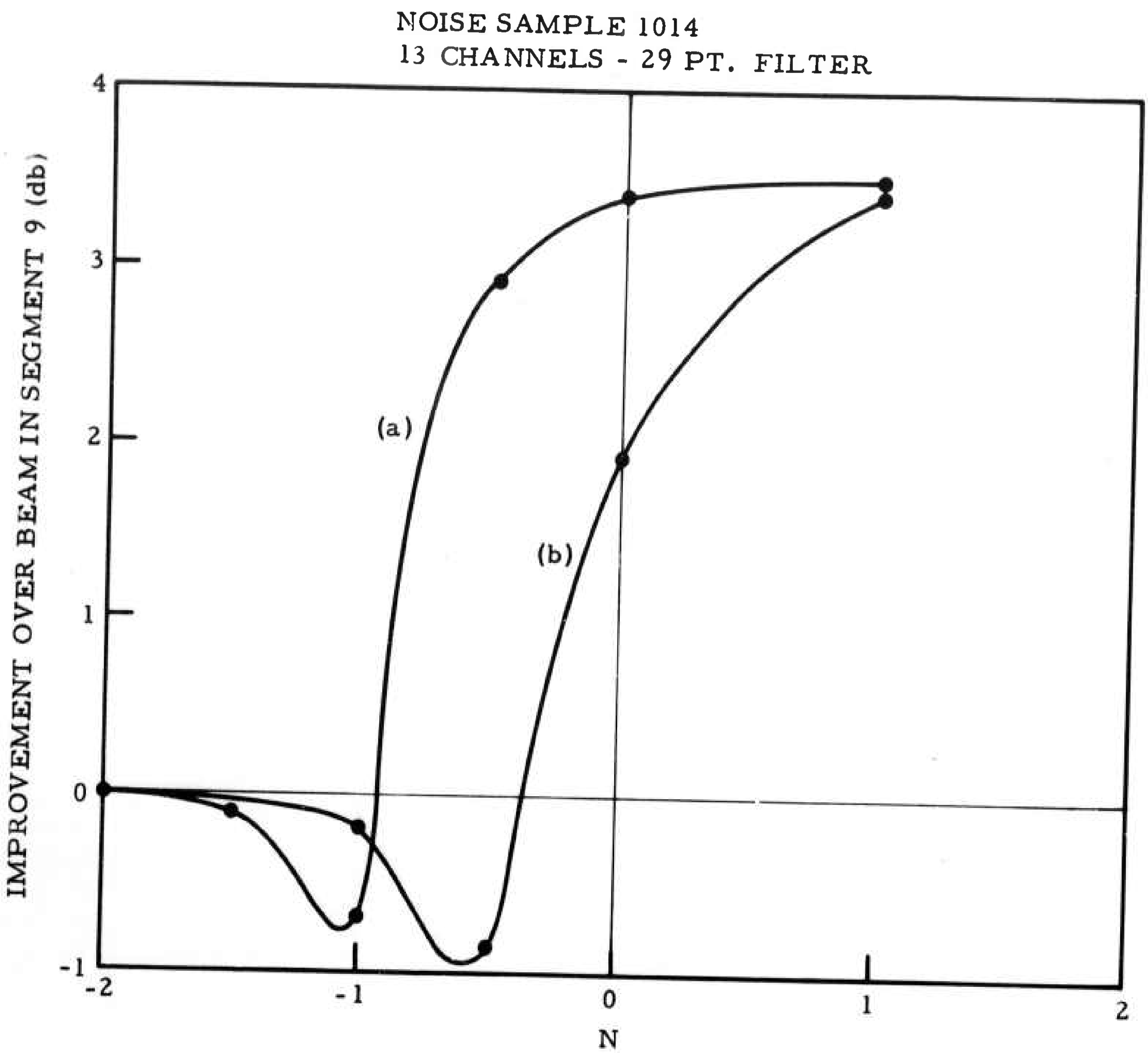
A noise sample consisting of 13 channels of data with 2560 points per channel was filtered with a 29 point filter. Initial filter coefficients were

$$F^{1}_{i,j} = \frac{\delta_{j,15.}}{NC} \qquad (8)$$

The filter output initially was, therefore, identical to the beam output but upon adaption with a K_s of 0.25 improvement of the filter output over the beam output was observed. The amount of this improvement for points 2049 through 2304 was calculated and has plotted in Figure 1 as a function of N.

For case (a) (Filter output compared to beam output) with a N value of 1 the filter was never frozen and the improvement of the filter output over the beam output in the 9^{th} of the 10 256 point data segments was 3.49 db. For smaller N values the filters began freezing on the noise as evidenced by the drop in filter improvement. For this case N=0 is the smallest acceptable value.

Similar processing for case (b) (filter output compared tc filter output) showed a minimum N value of 1 for this case. Case (b) was selected for use in subsequent processing for this task since in this case the minimum N value is not dependent on filter performance as it would be in case (a).



$$Y^t > (N+2)\,Q_t^{1/2}$$

where

(a) $Q_t = (1-\mu_2)\,\bar{X}_t^2 + \mu_2\, Q_{t-1}$

(b) $Q_t = (1-\mu_2)\,Y_t^2 + \mu_2\, Q_{t-1}$

$\mu_2 = 0.9$, $k_s = 0.25$

Figure 1. S/N Improvement as a Function of "Freezing" Threshold

Future plans for this task consist of embedding an infinite velocity signal in the noise sample at various S/N ratios. The data will then be processed with the case (b) signal detection criterion and a N value of 1. Results of this processing will demonstrate the effectiveness of the signal detection scheme in freezing the filters for the various signal sizes. Frequency-wavenumber responses of the filters will be computed before and after the signal so that the effects of the signal on the filter response can be measured.

Data Quality

Problems of data quality such as dead or noisy channels will certainly be encountered by an on-line processor. Thus techniques of handling these problems automatically must be developed. One such technique has been added to the IBM 360 on-line simulation program.

The program has been modified so that a channel will be deleted and its filter coefficients redistributed if it is found to be dead or noisy. A channel will be considered dead if

$$\left[\frac{1}{N}\sum_{n=0}^{N}(X_i^{t+n})^2\right]^{1/2} < \frac{XMIN}{NC}\sum_{i=1}^{NC}\left[\frac{1}{N}\sum_{n=0}^{N}(X_i^{t+n})^2\right]^{1/2} \tag{9}$$

or excessively noisy if

$$\left[\frac{1}{N}\sum_{n=0}^{N}(X_i^{t+n})^2\right]^{1/2} > \frac{XMAX}{NC}\sum_{i=1}^{NC}\left[\frac{1}{N}\sum_{n=0}^{N}(X_i^{t+n})^2\right]^{1/2} \tag{10}$$

where N, XMIN, and XMAX are variables read into the program.

If a channel meets either of the above conditions it is deleted and its filter coefficients are distributed according to equation (11).

$$
\begin{aligned}
F^{t}_{i,j} &= F^{t}_{i,j} + \frac{1}{NC - 1} F^{t}_{NDEL,j} \qquad i \neq NDEL \\
F^{t}_{i,j} &= 0
\end{aligned}
\tag{11}
$$

where NDEL is the number of the channel deleted.

Processing of data modified to include one or more dead or noisy channels is planned. Sufficient processing will be undertaken to establish the effectiveness of this type of data quality control and also to give a feel for the best values of the parameters N, XMIN, and XMAX.

Sampling Effects on Directional Beams

If the adaptive beam is to be directed for detection of signals other than infinite velocity, the sampling interval of the digital data will have some effect on the filter output. This would be expected because the signal model would have to be specified to the nearest sample point and will therefore not exactly match the actual signal. The larger the sample interval the larger this effect would be.

It is planned to investigate the magnitude of this problem processing the filter coefficients of a well adapted maximum-likelihood filter with a modified wavenumber response program. The wavenumber response program will perform the calculation

$$\rho\,(f,\vec{k}) = \sum_{i=1}^{NCH} \Phi_i\,(f)\; e^{-2\pi i\vec{k}\cdot(\vec{x}_i+\Delta\vec{x}_i)} \qquad (12)$$

where $\Delta\vec{x}_i = \frac{f}{\vec{k}}\,\Delta t_i$ $\qquad$ $\Phi_i(f)$ is the filter weight for the i^{th} channel

The increment Δx_i introduces into the calculation the effect of a time misalignment Δt_i in channel i. Comparison of the resulting wave-number responses will demonstrate the effects of sampling interval on directional adaptive beams.

Responses will be calculated for various sample rates and various $\vec{k}$ values > 8 km/sec.

SECTION IV: LONG-PERIOD ARRAY DATA ANALYSIS

A. TFO AND UBO LONG-PERIOD NOISE ANALYSIS

Two long-period noise samples from TFO and three from UBO were transformed from the time to the frequency domain using the Fast Fourier Transform technique. The segment length for each transform was 128 points (for 65 output frequencies) in the case of the TFO noise samples; 64 point transforms (for 33 output frequencies) were used for the shorter UBO noise samples. The program to perform the transform included provisions for omitting bad channels and segments of bad data, both of which were present in the noise samples.

The transforms output from this program were input to a crosspower matrix generation program which generated from each input transform a crosspower matrix for each output frequency. The resulting matrices were stacked at each frequency to form an average crosspower matrix at each frequency for the entire noise sample.

Using the crosspower matrices, multiple coherences were obtained in order to study the degree of predictability of a given element from other elements of each array. The coherence between the vertical component and the two horizontal components at each instrument location and between components at the center of the array and other like components of the array were determined.

The crosspower matrices were also used to compute two-dimensional f-k power density spectra for each set of like components for each noise sample. Both conventional and high resolution spectra were computed in each case.

The frequencies at which the f-k spectra were computed for display were chosen from regions of the single channel power density spectra which showed significant peaks or troughs.

Using azimuth intervals of ten degrees and wavenumber values corresponding to the Rayleigh wave velocity at the chosen frequency, the f-k power density program also computed for each chosen frequency the ratio of the array gain of the maximum-likelihood processor to the beamsteer processor.

Processing and analysis of the long-period noise data is essentially complete except for the investigation of an unusual low frequency energy peak which has been observed in the f-k spectra of the TFO data. The energy occurs at frequencies near 0.016 Hz and appears to propagate from the southwest at acoustic velocity. This feature is being analyzed in an attempt to determine its source. Upon completion of this investigation, a special report on the long-period noise analysis will be submitted for approval.

B. NEPENTHE

The Nepenthe process[3] is a single channel technique for suppressing stationary long-period noise.

Briefly, the process uses an estimator group of non-overlapping segments of long-period noise to estimate a noise amplitude spectrum, which is then subtracted from the amplitude spectrum of a segment to be treated. The difference amplitude spectrum (with negative values set equal to zero) and the unaltered phase spectrum of the original treated segment are used to transform back to a time-domain trace. The process then advances one segment and repeats.

To check the Nepenthe process and evaluate changes, a time trace was generated using a long-period signal from TFO and a noise recording from UBO combined at signal to noise ratios of 1:8, 1:4, 1:2, 1:1 and 4:1.

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(3) Simons R.S., W.R. Weber, and H.S. Travis, 1968: Preliminary Report on a Single Channel Statistical Technique for Suppressing Long-Period Microseismic Noise, Geotech Tech Rpt. 68-25, Contract F33657-68-C-0734, Project VT-8703, 19 July.

Two changes in the Nepenthe noise amplitude spectrum estimation were investigated.

- The original Nepenthe process uses the square root of the upper 90% limit of a scaled chi-square distribution as the estimated noise spectra. A change to the upper 75% limit was made. A comparison between the two spectrum show a difference of less than 1 db from 0 to 0.15 Hz which is the general signal range. The time-domain output of the two spectra are nearly identical.
- The calculation of the number of degrees of freedom using maximum-likelihood estimators for the scaled chi-square distribution is lengthy. In an effort to reduce this, a constant value for the number of degrees of freedom "b" was used which is twice the number of nonoverlapping segments used in the estimate.

For these data the value of "b" was 22 whereas the average value of "b" calculated by the Nepenthe process is 4.88 ± 0.26. The difference between the two methods is also less than 1 db from 0 to 0.15 Hz.

Also investigated was a change in the application of the estimated noise spectrum to the treated segment amplitude spectra. The original process is a subtraction. Rather than subtract the two, a comparison of amplitudes was made at each frequency. If the estimated noise was less than the treated segment, then the treated segment was passed entirely at that frequency. If the noise was greater, the treated segment amplitude was set to zero. This pass-no pass routine results in a higher amplitude output for the process, although more noise is admitted. The pass-no pass technique appears to give better results in the lower signal-to-noise ratios.

In order to evaluate the effect of the process on the signal, frequencies outside of a band from 0.02 to 0.06 Hz were suppressed to eliminate non-signal energy. A 0.02 to 0.06 Hz bandpass filter was applied to the original time traces to provide a comparison of the various processes.

Based on the few signals processed the Nepenthe process appears to be only marginally superior to a bandpass filter. Signal distortion is severe at low S/N ratio. The pass-no pass modification of the Nepenthe process eliminates some of the signal distortion.

This pass-no pass modification is a good frequency filtering scheme which should be included in a general package of long-period processing programs. Its usefulness is probably limited to regions where it is difficult to make matched filtering work effectively.

C. OVERLAPPING SIGNAL STUDY

Based on the duration of many surface waves it is highly probable that a surface wave from an event of special interest will sometimes be observed by the surface wavetrain from another event. Previous work[4] done on using multichannel linear filtering to separate such signals was not completed. Certain important questions remained unanswered.

An experiment done with data which was a compounding of Rayleigh Waves from a Hokkaido and a New Heberdies event was somewhat disappointing.

The vertical traces from A_0 and the C ring instruments were used and the New Heberdies Event was considered to be the desired signal and added in at about 1/10 the amplitude of the Hokkaido Event. This compound data was then used to measure the correlation structure of the "noise" and a signal model with time delays corresponding to the New Hiberdies event was generated.

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(4) Multicomponent Long-Period Signal Separation Advanced Array Research Report No. 6, Texas Instruments Incorporated, 28 April 1969.

The resulting Wiener Filter set was able to suppress the Hokkaido Event by only about 16 db[4]. It is possible that because the measured "noise" correlation included both events that the suppression of the interferring event may be proportional to the signal-to-noise ratio. If this is the case then a multichannel Wiener Filter, followed by a matched filter, should give a useful estimate of the subject event strength over a substantial range of magnitude differences.

The experiment is being repeated with a noise-to-signal ratio of 40 db and ∞. If these results are encouraging a standard procedure for interferring signal separation will be recommended.

The necessary programs for this experiment have been collected and are being checked out. No results have been obtained to date.

In addition, 3-component Wiener Filtering techniques will be investigated further. In general there was poor coherence among horizontal and vertical components of the Rayleigh waves at LASA[4]. These coherences will be checked at UBO. If they are encouraging, some 3-component signal separation filters will be evaluated using data recorded on the UBO long-period array.

D. MATCHED FILTERING STUDY

Using 17 events from the Kuriles-Kamchatka region recorded at LASA a study was conducted to determine the effectiveness of various matching waveforms for doing matched filtering of Rayleigh Waves. Chirp waveforms, waveforms derived from average crustal models, and master events from within the region were used as matching waveforms. These data were analyzed to give some guidance to initial routine surface wave processing at the SAAC Center.

For the signals studied from the Kuriles, routine matched filtering at LASA does not appear to be a very powerful tool. Two routine methods of operation appear reasonable.

Since the output of the matched filtering operation can be used directly as a discriminant related to the total energy in the wavetrain[5] (in a fixed region at least), it would be desirable to make this measurement using a matching filter which would give a high correlation coefficient with most events. The signals analyzed indicate that this would be accomplished by using only about 3/4 of the expected duration of a simple event and employing either a fairly close (800 km) master event or a simple chirp waveform. Both of these gave average correlation coefficients on the order of 0.8. There were not sufficient signals processed to give a good estimate of the variability of such a procedure, but an assumption of equal correlation with each event does not appear to be unreasonable.

If it is desired to improve upon the above procedure it would require a close (within about 200 km) master event. Using such a master event and the entire, or nearly the entire, wavetrain, a small S/N improvement (a factor of about 1.7 or 2.3 db) was obtained. The average correlation coefficient was 0.755 for 512 seconds using master events within 200 km. This is comparable to what could be obtained using a chirp wave form and 300 seconds (0.77). Again, there was insufficient data to make a really meaningful estimate of the variability of these correlation coefficients.

Special Scientific Report No. 4 "A Preliminary Study of Techniques for Routine Matched Filtering of Surface Waves" which details this work was submitted for approval during this past quarter.

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(5) Seismic Data Laboratory Report No. 175, 1967, Detection of surface waves from small events at teleseismic distances.

SECTION V: TFO EXTENDED SHORT-PERIOD ARRAY NOISE STUDY

A study of the ambient noise field for the TFO extended short-period seismic array has been conducted on data recorded during two periods designated as 'Winter' and 'Summer'. The objective was to compare the seismic noise field between the two periods and determine the effectiveness of multichannel processing using the extended array. Preparation of a special report on this study is near completion. A brief summary of the analysis and MCF evaluation conducted on the data is presented in the following.

A. AMBIENT NOISE ANALYSIS

Single-channel power density spectra were computed for various elements of the extended short-period array. Using the daily calibrations, absolute levels of the spectra at 1.0 Hz were established. A comparison of the spectra for a single channel, as shown in Figure 2, shows a significant change in the absolute level of the noise field over a period of time. An analysis of atmospheric pressure maps indicate that the increase in noise level is directly related to low pressure activity.

High resolution wavenumber spectra at the frequency showing the large increase in noise power (.12 Hz) in the 'Winter' data is shown in Figure 3. A surface mode source from the Northwest, the direction of intense pressure activity near the Alaskan coast, is observed in the spectra. Significant P-wave energy is also observed from this generating source at frequencies up to .45 Hz. At higher frequencies the Northwest lobe is not so dominating and several surface mode sources can be detected.

In the summer data, surface mode energy from the Northeast is predominate at the lower frequencies. However, in the frequency area of .4 Hz and higher the surface mode energy becomes very isotropic for both noise samples. (Figure 4).

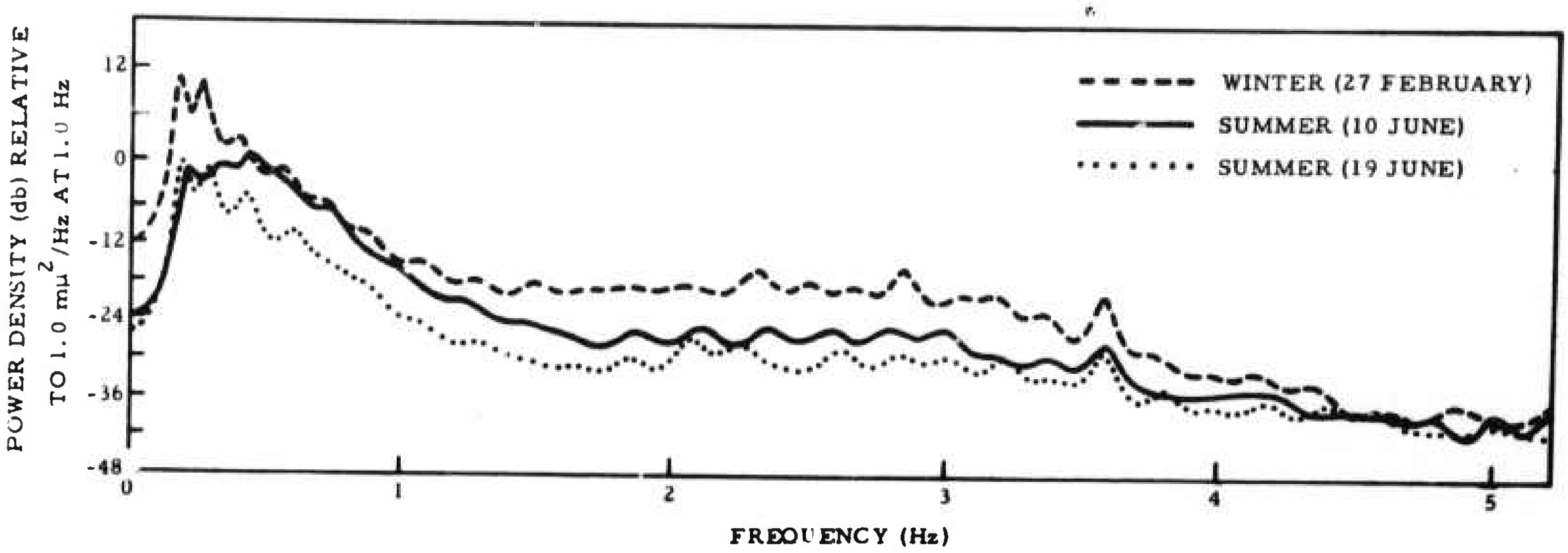

Figure 2. Comparison of Single-Channel Power Density Spectra from Various Time Periods

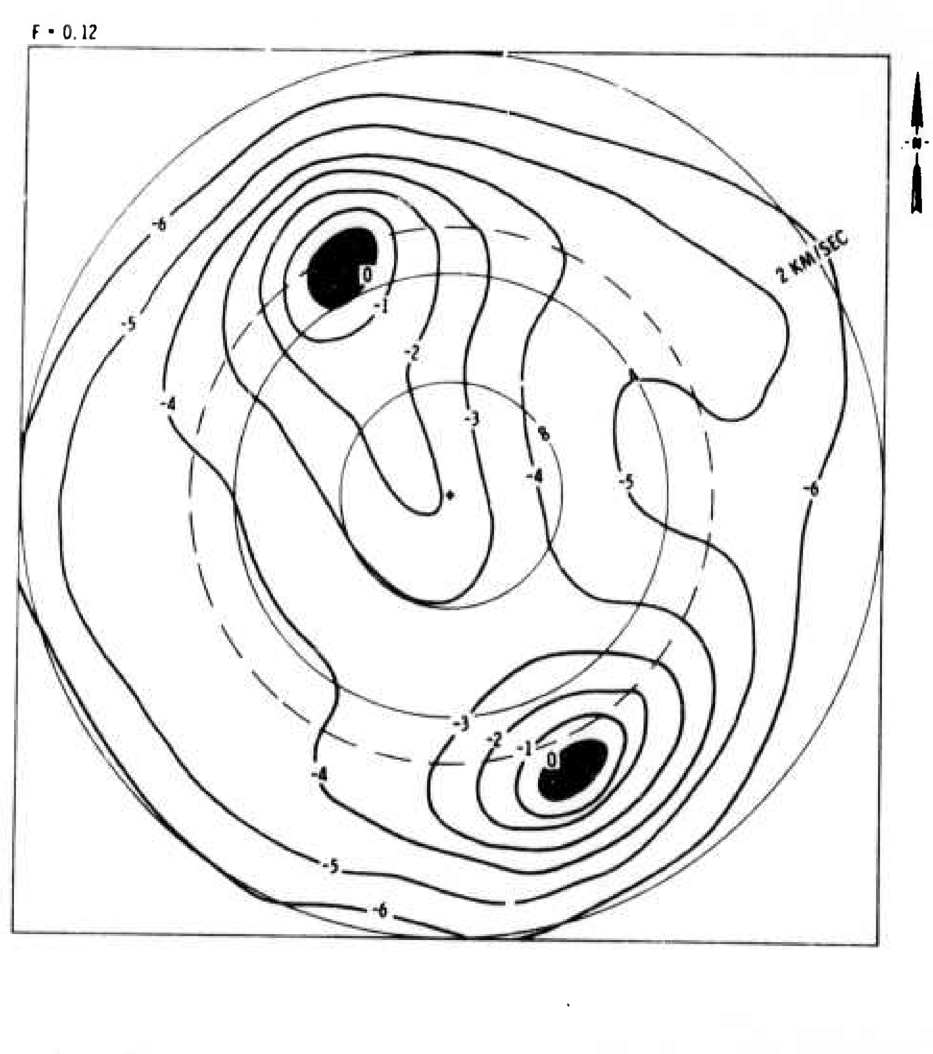

Figure 3. Wavenumber Spectra of 'Winter .12 Hz Noise

The primary difference in the P-wave structure of the two noise samples exist at frequencies below .45 Hz. In the 'Summer' data, the P-wave energy from the south with an extremely high apparent horizontal velocity is the predominant contribution to the P-wave noise field. However, in the 'Winter' data, the source to the Northwest of the array provide the most significant contribution to the P-wave noise field at .45 Hz and lower. (Figure 5). Above. 45 Hz the wavenumber spectra of the P-wave noise are quite similiar for both periods of time. Both show multiple number of P-wave contributors, with the principle contributor from the south with a very high apparent horizontal velocity. (Figure 6)

From K-line spectra computed using the three arms of the extended array, an estimate of the percentage of the total noise field for which each of the defined noise lobes of the wavenumber spectra are responsible was made. These percentages for the 4 directional surface mode lobes, the P-wave energy, and isotropic surface mode energy is illustrated in Figure 7.

B. MCF DESIGN AND EVALUATION

A set of multichannel filters were designed from the 'Winter' and the 'Summer' data. The set of MCFs for each period consist of 3 MCFs, each one for a particular subset of the extended array elements. That is, a MCF was designed using the center element and the intermost ring; another, using the same as the preceding plus the next ring; and a third, using all of the available elements.

Evaluation of the MCFs was obtained by applying the frequency domain designed MCFs to the noise and signal power matrices from which they were designed. A similiar evaluation of straight summation processing was made on the design matrices.

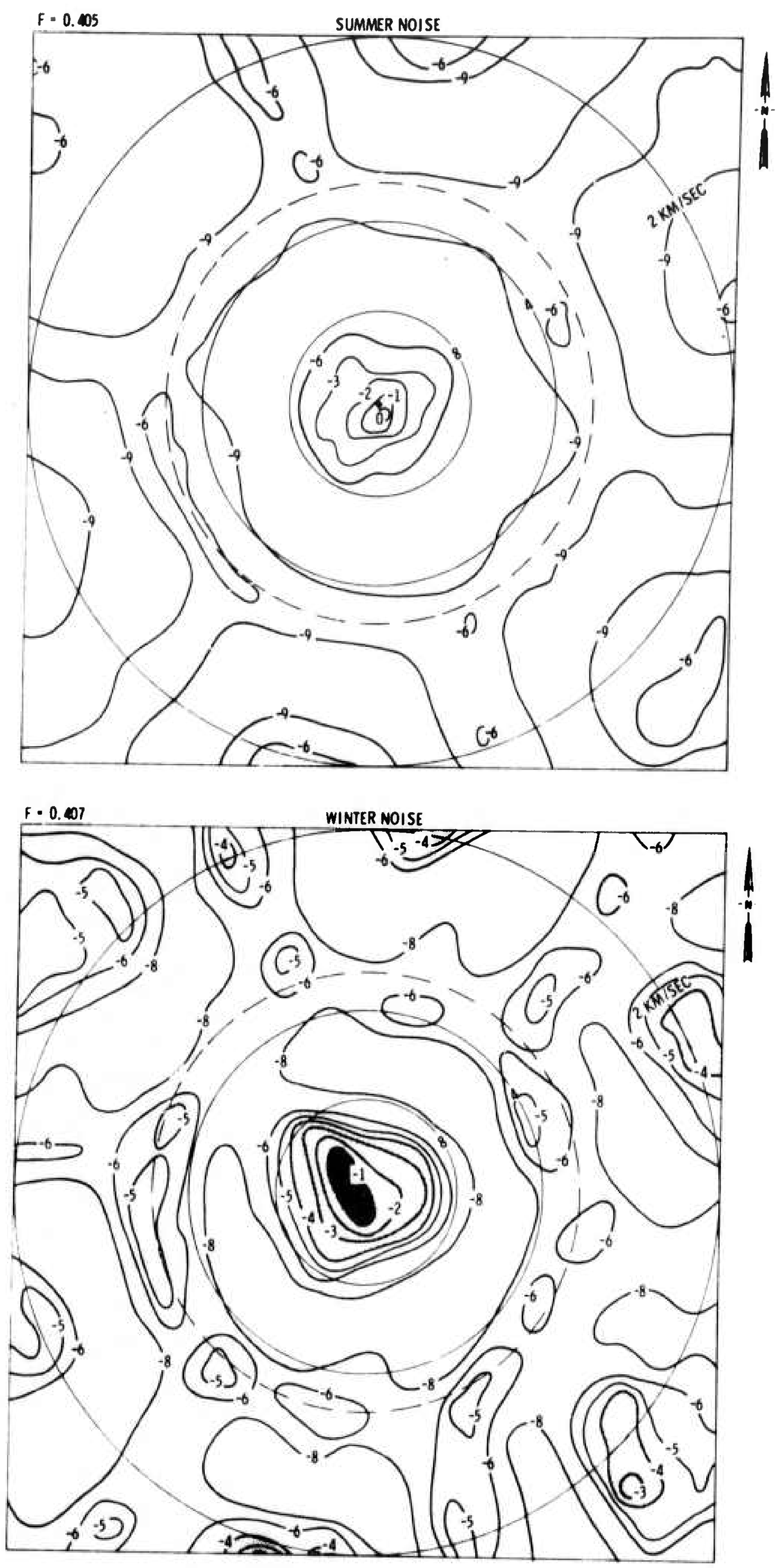

Figure 4. Wavenumber Spectra of .4 Hz Energy Showing Isotropic Character of Noise Field

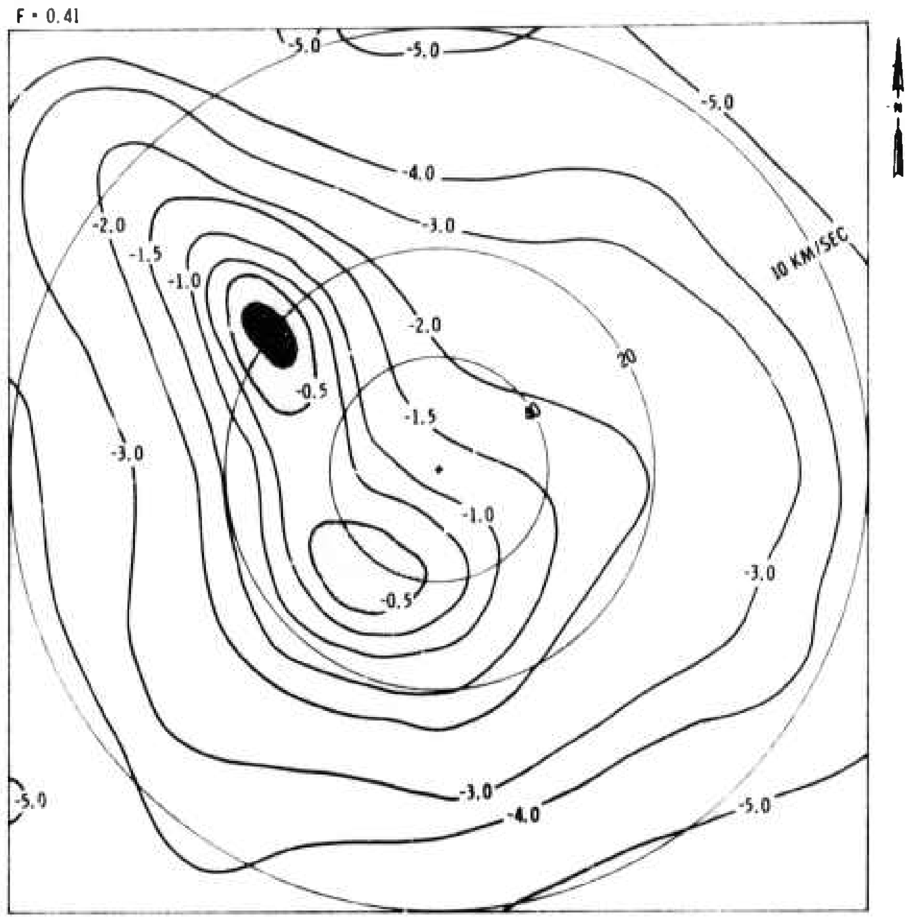

Figure 5. Wavenu› oer Spectra of P-Wave Noise from 'Winter' Noise Sample

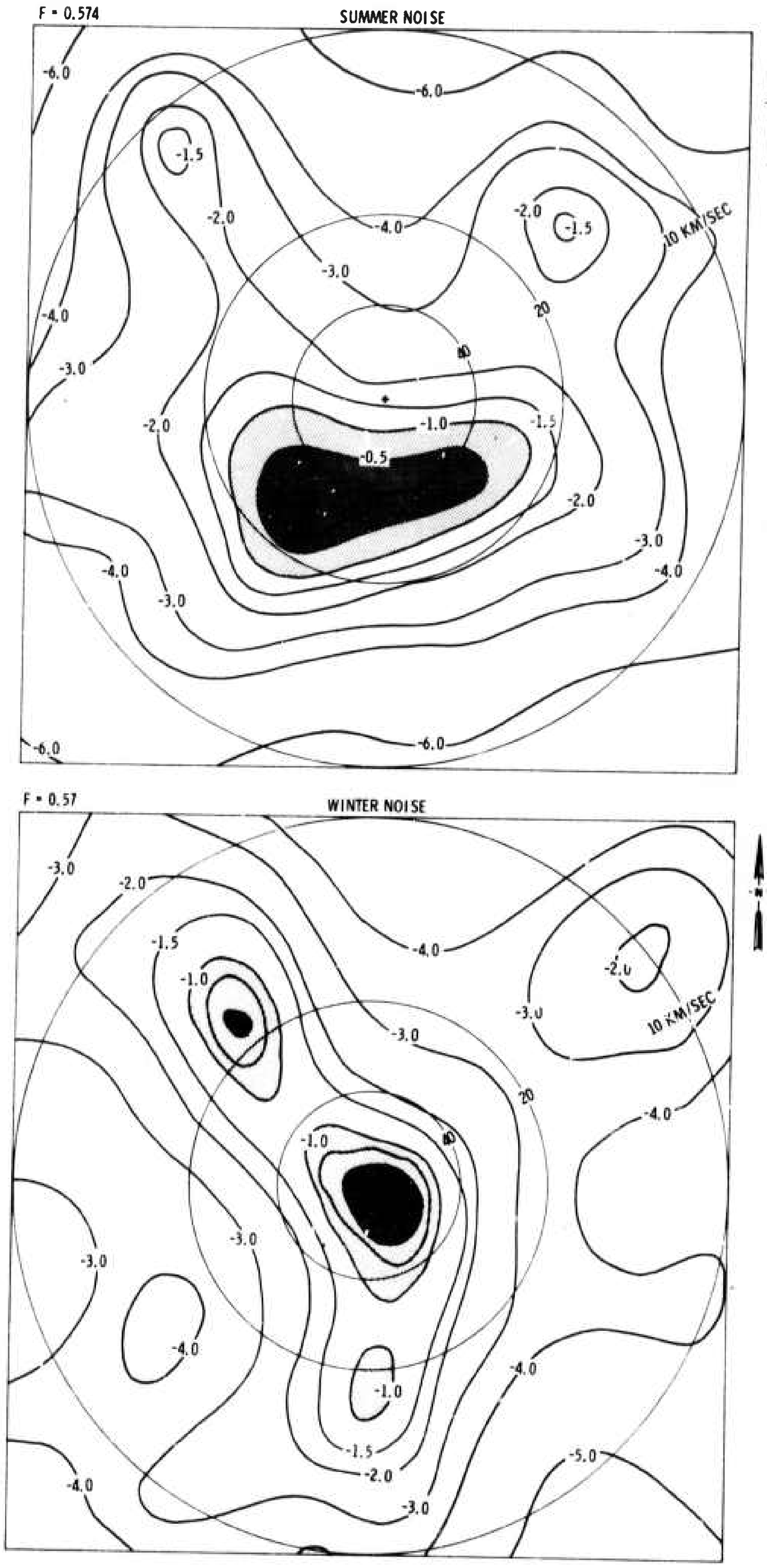

Figure 6. Wavenumber Spectra of 'Summer' and 'Winter' P-Wave Noise at .57 Hz

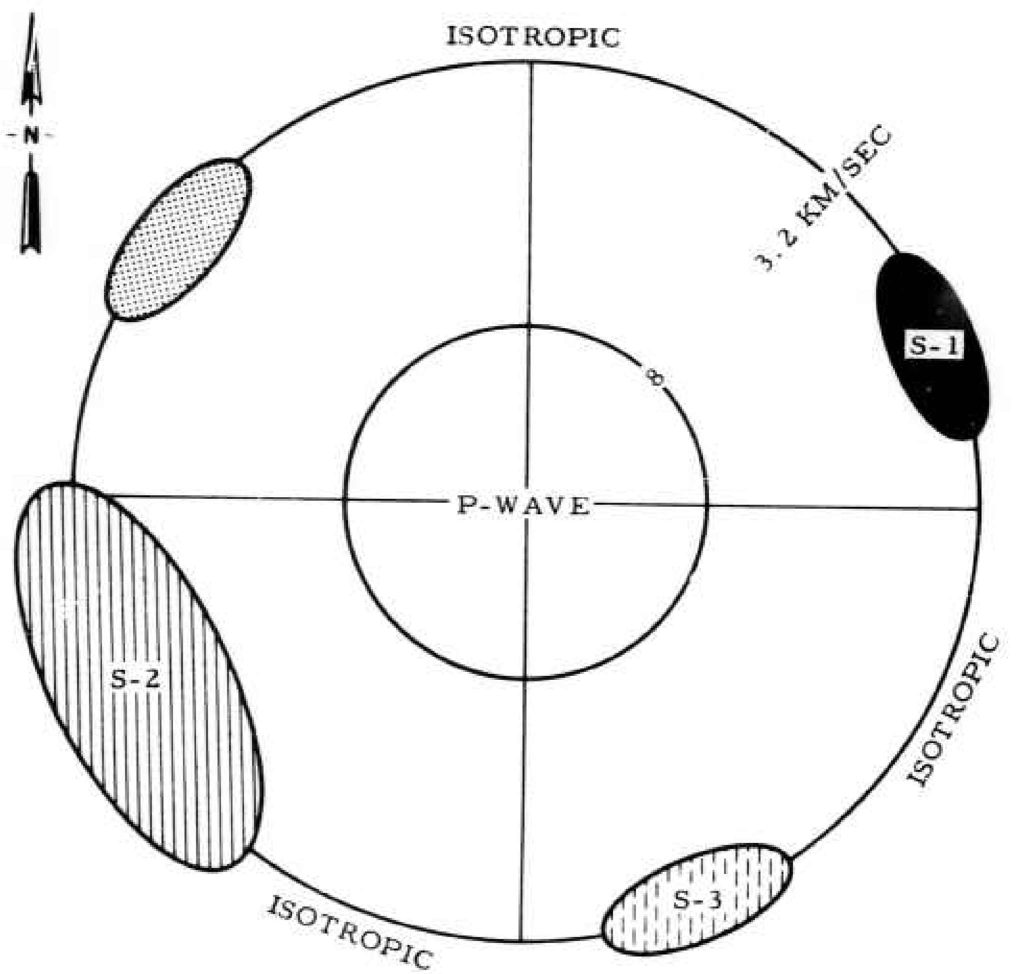

ILLUSTRATION OF WAVENUMBER
SPACE REFERENCED FOR EACH
PRINCIPLE CONTRIBUTOR

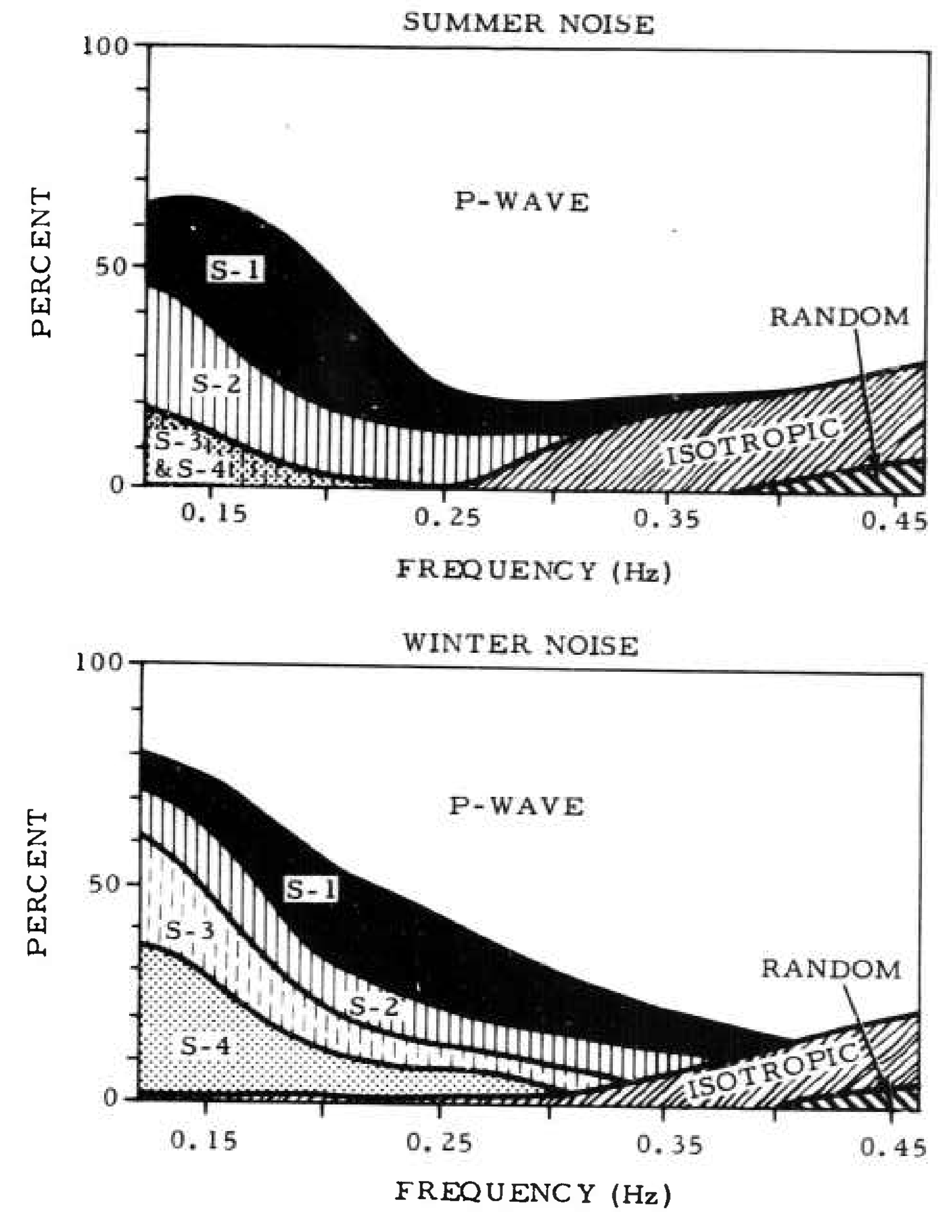

Figure 7. Distribution of Ambient Noise Power for Identified Sources.

Comparison of the MCF processing to the simple summation process (Figure 8) shows that very little if any signal-to-noise improvement can be obtained by MCF processing over that of beamsteering at frequencies above .75 Hz. Some additional signal-to-noise improvement is obtained by MCF processing at lower frequencies, especially in the 'Winter' noise where a significant percentage of the energy is due to the surface-mode energy from the pressure activity off the Alaskan coast. However, since this energy will not be time or space stationary, fixed MCF processing of the extended array would not provide continued improvement. Therefore, based on this evaluation, MCF processing using fixed filters on the extended array would not provide any appreciable improvement over beamsteering.

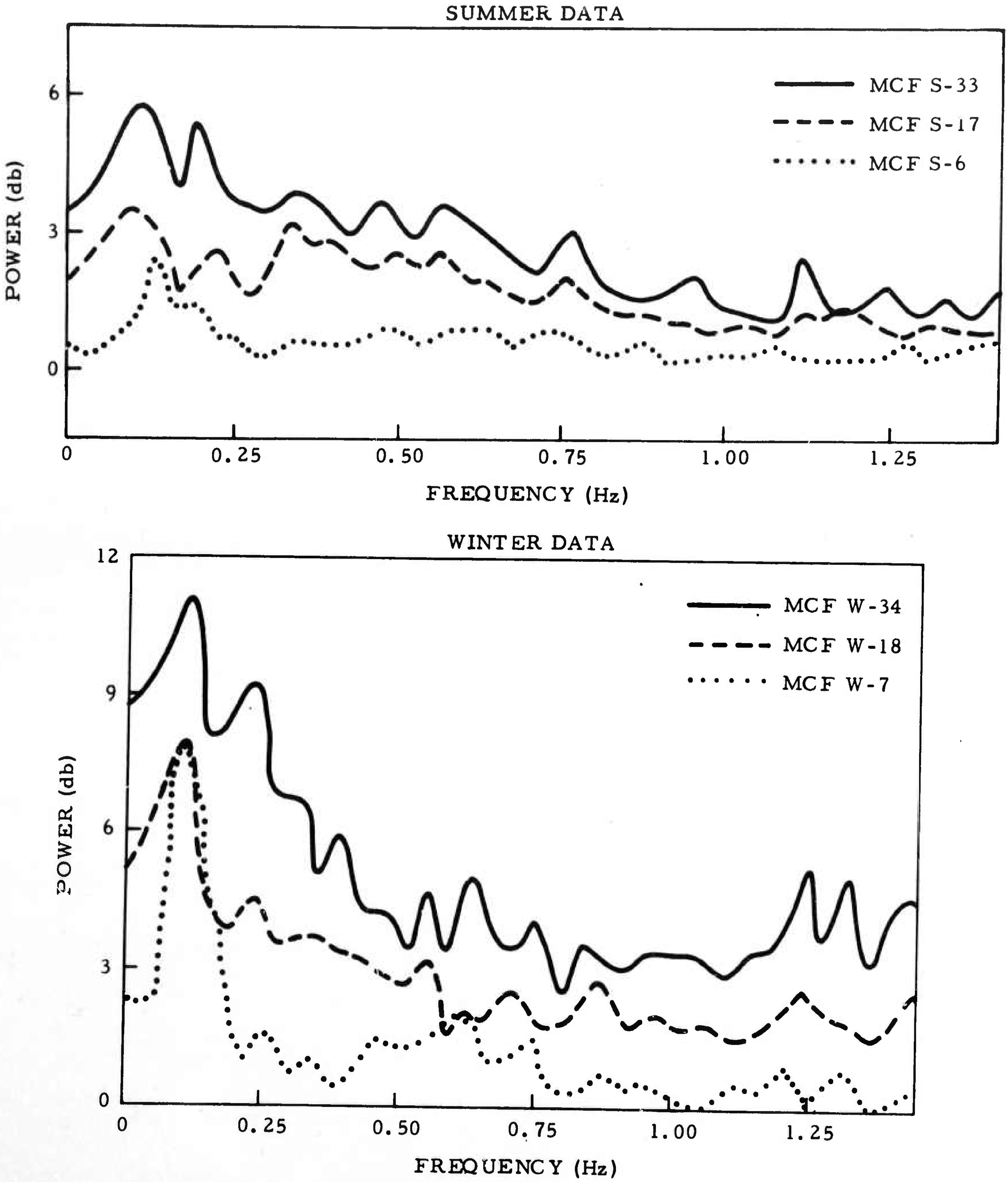

Figure 8. Signal-To-Noise Improvement of MCF Processing of TFO Extended Array Relative to Beamsteer Processing

SECTION VI: HI RESOLUTION WAVENUMBER SPECTRA DISPLAYS

The goal of the high resolution spectra program is to simulate a real time, continuously generated, high resolution f-k spectrum for both long- and short-period data and to output the spectrum as a contoured plot. A high resolution frequency-wavenumber (f-k) program to produce f-k spectra plots has been written and debugged. The program has been written to include a wide range of possible processing procedures including a beamsteered f-k plot. The time gate for consecutive plots has been overlapped by 50%.

A list of the options in the program and other input parameters with their limitations are:

- Option to stack f-k plots over frequency up to five frequencies
- Will process up to 20 channels
- Option to apply Hanning function to input
- Two velocity ranges on output plots (in km/sec)

  $8 \leq V < \infty$ for short-period data
  $2 \leq V < \infty$ for long-period data
- Decay factor in crosspower matrix update is codeable
- Crosspower matrix may be prewhitened a selected amount
- Option to generate and update crosspower matrix for a period of time before producing output plots

The program is presently being run on long-period data from UBO.

SECTION VII: REFERENCES

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(5) Seismic Data Laboratory Report No. 175, 1967, Detection of surface waves from small events at teleseismic distances.

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13. ABSTRACT

This third quarterly report discusses work progress and summarizes results obtained. The work categories are: Design of fixed multichannel filters off-line; on-line adaptive processing; long-period array data analysis; short-period TFO array data analysis; and Hi resolution wavenumber spectra displays of seismic array data.

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14. KEY WORDS	LINK A		LINK B		LINK C	
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